



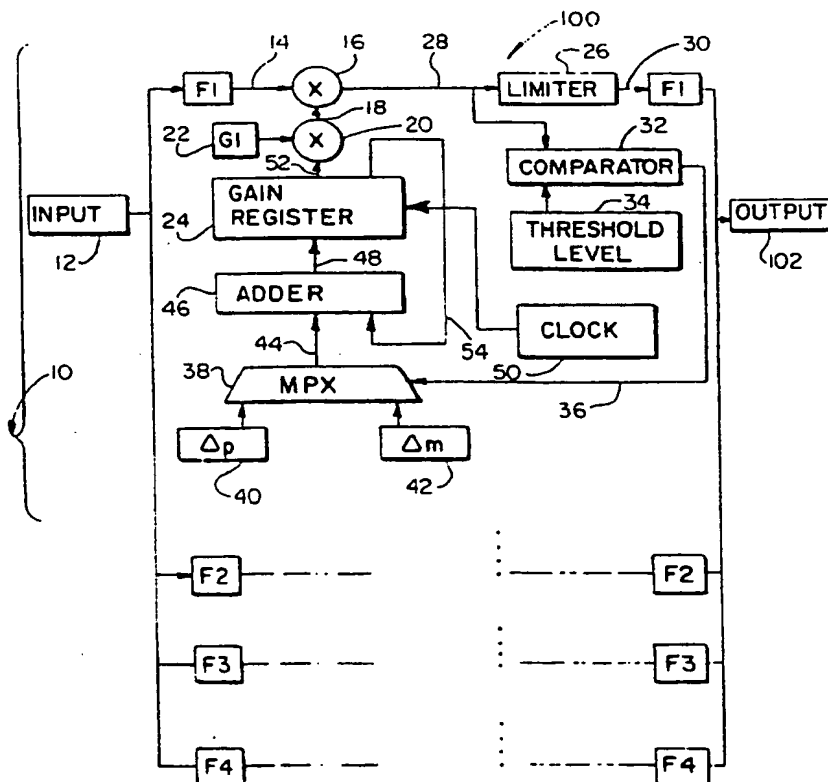
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(54) Title: **ADAPTIVE GAIN AND FILTERING CIRCUIT FOR A SOUND REPRODUCTION SYSTEM**

(57) Abstract

Adaptive compressive gain and level dependent spectral shaping circuitry for a hearing aid include a microphone to produce an input signal and a plurality of channels connected to a common circuit output (102). Each channel has a preset frequency response. Each channel includes a filter (F1, F2, F3, F4) with a preset frequency response to receive the input signal (12) and to produce a filtered signal, a channel amplifier to amplify the filtered signal to produce a channel output signal, a threshold register (34) to establish a channel threshold level, and a gain circuit (24). The gain circuit increases the gain of the channel amplifier when the channel output signal falls below the channel threshold level and decreases the gain of the channel amplifier when the channel output signal rises above the channel threshold level. A transducer produces sound in response to the signal passed by the common circuit output.



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ADAPTIVE GAIN AND FILTERING CIRCUIT
FOR A SOUND REPRODUCTION SYSTEM

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Background of the Invention

20 The present invention relates to adaptive compressive gain and level dependent spectral shaping circuitry for a sound reproduction system and, more particularly, to such circuitry for a hearing aid.

 The ability to perceive speech and other sounds 25 over a wide dynamic range is important for employment and daily activities. When a hearing impairment limits a person's dynamic range of perceptible sound, incoming sound falling outside of the person's dynamic range should be modified to fall within the limited dynamic 30 range to be heard. Soft sounds fall outside the limited dynamic range of many hearing impairments and must be amplified above the person's hearing threshold with a

hearing aid to be heard. Loud sounds fall within the limited dynamic range of many hearing impairments and do not require a hearing aid or amplification to be heard. If the gain of the hearing aid is set high enough to enable perception of soft sounds, however, intermediate and loud sounds will be uncomfortably loud. Because speech recognition does not increase over that obtained at more comfortable levels, the hearing-impaired person will prefer a lower gain for the hearing aid. However, a lower gain reduces the likelihood that soft sounds will be amplified above the hearing threshold. Modifying the operation of a hearing aid to reproduce the incoming sound at a reduced dynamic range is referred to herein as compression.

It has also been found that the hearing-impaired prefer a hearing aid which varies the frequency response in addition to the gain as sound level increases. The hearing-impaired may prefer a first frequency response and a high gain for low sound levels, a second frequency response and an intermediate gain for intermediate sound levels, and a third frequency response and a low gain for high sound levels. This operation of a hearing aid to vary the frequency response and the gain as a function of the level of the incoming sound is referred to herein as "level dependent spectral shaping."

In addition to amplifying and filtering incoming sound effectively, a practical ear-level hearing aid design must accommodate the power, size and microphone placement limitations dictated by current commercial hearing aid designs. While powerful digital signal processing techniques are available, they can require considerable space and power so that most are not suitable for use in an ear-level hearing aid. Accordingly, there is a need for a hearing aid that varies its gain and frequency response as a function of the level of incoming sound, i.e., that provides an

adaptive compressive gain feature and a level dependent spectral shaping feature each of which operates using a modest number of computations, and thus allows for the customization of variable gain and variable filter parameters according to a user's preferences.

Summary of the Invention

Among the several objects of the present invention may be noted the provision of a circuit in which the gain is varied in response to the level of an incoming signal; the provision of a circuit in which the frequency response is varied in response to the level of an incoming signal; the provision of a circuit which adaptively compresses an incoming signal occurring over a wide dynamic range into a limited dynamic range according to a user's preference; the provision of a circuit in which the gain and the frequency response are varied in response to the level of an incoming signal; and the provision of a circuit which is small in size and which has minimal power requirements for use in a hearing aid.

Generally, in one form the invention provides an adaptive compressing and filtering circuit having a plurality of channels connected to a common output. Each channel includes a filter with preset parameters to receive an input signal and to produce a filtered signal, a channel amplifier which responds to the filtered signal to produce a channel output signal, a threshold circuit to establish a channel threshold level for the channel output signal, and a gain circuit. The gain circuit responds to the channel output signal and the channel threshold level to increase the gain setting of the channel amplifier up to a predetermined limit when the channel output signal falls below the channel threshold level and to decrease the gain setting of the channel amplifier when the channel output signal rises above the channel threshold level. The channel output signals are

combined to produce an adaptively compressed and filtered output signal. The circuit is particularly useful when incorporated in a hearing aid. The circuit would include a microphone to produce the input signal and a transducer to produce sound as a function of the adaptively compressed and filtered output signal. The circuit could also include a second amplifier in each channel which responds to the filtered signal to produce a second channel output signal. The hearing aid may additionally include a circuit for programming the gain setting of the second channel amplifier as a function of the gain setting of the first channel amplifier.

Another form of the invention is an adaptive gain amplifier circuit having an amplifier to receive an input signal in the audible frequency range and to produce an output signal. The circuit includes a threshold circuit to establish a threshold level for the output signal. The circuit further includes a gain circuit which responds to the output signal and the threshold level to increase the gain of the amplifier up to a predetermined limit in increments having a magnitude dp when the output signal falls below the threshold level and to decrease the gain of the amplifier in decrements having a magnitude dm when the output signal rises above the threshold level. The output signal is compressed as a function of the ratio of dm over dp to produce an adaptively compressed output signal. The circuit is particularly useful in a hearing aid. The circuit may include a microphone to produce the input signal and a transducer to produce sound as a function of the adaptively compressed output signal.

Still another form of the invention is a programmable compressive gain amplifier circuit having a first amplifier to receive an input signal in the audible frequency range and to produce an amplified signal. The circuit includes a threshold circuit to establish a

threshold level for the amplified signal. The circuit further includes a gain circuit which responds to the amplified signal and the threshold level to increase the gain setting of the first amplifier up to a predetermined limit when the amplified signal falls below the threshold level and to decrease the gain setting of the first amplifier when the amplified signal rises above the threshold level. The amplified signal is thereby compressed. The circuit also has a second amplifier to receive the input signal and to produce an output signal. The circuit also has a gain circuit to program the gain setting of the second amplifier as a function of the gain setting of the first amplifier. The output signal is programmably compressed. The circuit is useful in a hearing aid. The circuit may include a microphone to produce the input signal and a transducer to produce sound as a function of the programmably compressed output signal.

Still another form of the invention is an adaptive filtering circuit having a plurality of channels connected to a common output, each channel including a filter with preset parameters to receive an input signal in the audible frequency range to produce a filtered signal and an amplifier which responds to the filtered signal to produce a channel output signal. The circuit includes a second filter with preset parameters which responds to the input signal to produce a characteristic signal. The circuit further includes a detector which responds to the characteristic signal to produce a control signal. The time constant of the detector is programmable. The circuit also has a log circuit which responds to the detector to produce a log value representative of the control signal. The circuit also has a memory to store a preselected table of log values and gain values. The memory responds to the log circuit to select a gain value for each of the amplifiers in the

channels as a function of the produced log value. Each of the amplifiers in the channels responds to the memory to separately vary the gain of the respective amplifier as a function of the respective selected gain value. The
5 channel output signals are combined to produce an adaptively filtered output signal. The circuit is useful in a hearing aid. The circuit may include a microphone to produce the input signal and a transducer to produce sound as a function of the adaptively filtered output
10 signal.

Yet still another form of the invention is an adaptive filtering circuit having a filter with variable parameters to receive an input signal in the audible frequency range and to produce an adaptively filtered
15 signal. The circuit includes an amplifier to receive the adaptively filtered signal and to produce an adaptively filtered output signal. The circuit additionally has a detector to detect a characteristic of the input signal and a controller which responds to the detector to vary
20 the parameters of the variable filter and to vary the gain of the amplifier as functions of the detected characteristic.

Other objects and features will be in part apparent and in part pointed out hereinafter.

25 Brief Description of the Drawings

Fig. 1 is a block diagram of an adaptive compressive gain circuit of the present invention.

Fig. 2 is a block diagram of an adaptive compressive gain circuit of the present invention wherein
30 the compression ratio is programmable.

Fig. 3 is a graph showing the input/output curves for the circuit of Fig. 2 using compression ratios ranging from 0-2.

Fig. 4 shows a four channel level dependent spectral shaping circuit wherein the gain in each channel is adaptively compressed using the circuit of Fig. 1.

Fig. 5 shows a four channel level dependent spectral shaping circuit wherein the gain in each channel is adaptively compressed with a programmable compression ratio using the circuit of Fig. 2.

Fig. 6 shows a four channel level dependant spectral shaping circuit wherein the gain in each channel is adaptively varied with a level detector and a memory.

Fig. 7 shows a level dependant spectral shaping circuit wherein the gain of the amplifier and the parameters of the filters are adaptively varied with a level detector and a memory.

Fig. 8 shows a two channel version of the four channel circuit shown in Fig. 6.

Fig. 9 shows the output curves for the control lines leading from the memory of Fig. 8 for controlling the amplifiers of Fig. 8.

Detailed Description of Preferred Embodiments

An adaptive filtering circuit of the present invention as it would be embodied in a hearing aid is generally indicated at reference number 10 in Fig. 1. Circuit 10 has an input 12 which represents any conventional source of an input signal such as a microphone, signal processor, or the like. Input 12 also includes an analog to digital converter (not shown) for analog input signals if circuit 10 is implemented with digital components. Likewise, input 12 includes a digital to analog converter (not shown) for digital input signals if circuit 10 is implemented with analog components.

Input 12 is connected by a line 14 to an amplifier 16. The gain of amplifier 16 is controlled via a line 18 by an amplifier 20. Amplifier 20 amplifies the

value stored in a gain register 24 according to a predetermined gain setting stored in a gain register 22 to produce an output signal for controlling the gain of amplifier 16. The output signal of amplifier 16 is
5 connected by a line 28 to a limiter 26. Limiter 26 peak clips the output signal from amplifier 16 to provide an adaptively clipped and compressed output signal at output 30 in accordance with the invention, as more fully described below. The output 30, as with all of the
10 output terminals identified in the remaining Figs. below, may be connected to further signal processors or to drive the transducer (not shown) of a hearing aid.

With respect to the remaining components in circuit 10, a comparator 32 monitors the output signal
15 from amplifier 16 via line 28. Comparator 32 compares the level of said output with a threshold level stored in a register 34 and outputs a comparison signal via a line 36 to a multiplexer 38. When the level of the output signal of amplifier 16 exceeds the threshold level stored
20 in register 34, comparator 32 outputs a high signal via line 36. When the level of the output of amplifier 16 falls below the threshold level stored in register 34, comparator 32 outputs a low signal via line 36. Multiplexer 38 is also connected to a register 40 which
25 stores a magnitude dp and to a register 42 which stores a magnitude dm . When multiplexer 38 receives a high signal via line 36, multiplexer 38 outputs a negative value corresponding to dm via a line 44. When multiplexer 38 receives a low signal via line 36, multiplexer 38 outputs
30 a positive value corresponding to dp via line 44. An adder 46 is connected via line 44 to multiplexer 38 and is connected via a line 54 to gain register 24. Adder 46 adds the value output by multiplexer 38 to the value stored in gain register 24 and outputs the sum via a line
35 48 to update gain register 24. The circuit components for updating gain register 24 are enabled in response to

a predetermined portion of a timing sequence produced by a clock 50. Gain register 24 is connected by a line 52 to amplifier 20. The values stored in registers 22 and 24 thereby control the gain of amplifier 20. The output
5 signal from amplifier 20 is connected to amplifier 16 for increasing the gain of amplifier 16 up to a predetermined limit when the output level from amplifier 16 falls below the threshold level stored in register 34 and for
10 decreasing the gain of amplifier 16 when the output level from amplifier 16 rises above the threshold level stored in register 34.

In one preferred embodiment, gain register 24 is a 12 bit register. The six most significant bits are connected by line 52 to control the gain of amplifier 16.
15 The six least significant bits are updated by adder 46 via line 48 during the enabling portion of the timing sequence from clock 50. The new values stored in the six least significant bits are passed back to adder 46 via line 54. Adder 46 updates the values by dm or dp under
20 the control of multiplexer 38. When the six least significant bits overflow the first six bits of gain register 24, a carry bit is applied to the seventh bit of gain register 24, thereby incrementing the gain setting of amplifier 20 by one bit. Likewise, when the six least
25 significant bits underflow the first six bits of gain register 24, the gain setting of amplifier 20 is decremented one bit. Because the magnitudes dp and dm are stored in log units, the gain of amplifier 16 is increased and decreased by a constant percentage. A one
30 bit change in the six most significant bits of gain register 24 corresponds to a gain change in amplifier 16 of approximately $\frac{1}{4}$ dB. Accordingly, the six most significant bits in gain register 24 provide a range of 32 decibels over which the conditions of adaptive
35 limiting occur.

The sizes of magnitudes d_p and d_m are small relative to the value corresponding to the six least significant bits in gain register 24. Accordingly, there must be a net contribution of positive values corresponding to d_p in order to raise the six least significant bits to their full count, thereby incrementing the next most significant bit in gain register 24. Likewise, there must be a net contribution of negative values corresponding to d_m in order for the six least significant bits in gain register 24 to decrement the next most significant bit in gain register 24. The increments and decrements are applied as fractional values to gain register 24 which provides an averaging process and reduces the variance of the mean of the gain of amplifier 16. Further, since a statistical average of the percent clipping is the objective, it is not necessary to examine each sample. If the signal from input 12 is in digital form, clock 50 can operate at a frequency well below the sampling frequency of the input signal. This yields a smaller representative number of samples. For example, the sampling frequency of the input signal is divided by 512 in setting the frequency for clock 50 in Fig. 1.

In operation, circuit 10 adaptively adjusts the channel gain of amplifier 16 so that a constant percentage clipping by limiter 26 is achieved over a range of levels of the signal from input 12. Assuming the input signal follows a Laplacian distribution, it is modeled mathematically with the equation:

$$p(x) = 1/(\sqrt{2}R) e^{-(\sqrt{2}|x|/R)} \quad (1)$$

In equation (1), R represents the overall root means square signal level of speech. A variable F_L is now defined as the fraction of speech samples that fall outside of the limits $(L, -L)$. By integrating the

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Laplacian distribution over the intervals $(-\infty, -L)$ and $(L, +\infty)$, the following equation for F_L is derived:

$$F_L = e^{-(\sqrt{2})L/R} \quad (2)$$

As above, when a sample of the signal from input 12 is in the limit set by register 34, the gain setting in gain register 24 is reduced by dm . When a sample of the signal from input 12 is not in limit, the gain is increased by dp . Therefore, circuit 10 will adjust the gain of amplifier 16 until the following condition is met:

$$(1-F_L)dp = F_L dm \quad (3)$$

After adaption, the following relationships are found:

$$dp = F_L(dp + dm) \quad (4)$$

$$R/L = \sqrt{2}/\ln(1 + dm/dp) \quad (5)$$

Within the above equations, the ratio R/L represents a compression factor established by the ratio dm/dp . The percentage of samples that are clipped at $\pm L$ is given by:

$$\% \text{ clipping} = F_L * 100 \quad (6)$$

Table I gives typical values that have been found useful in a hearing aid. Column three is the "headroom" in decibels between the root mean square signal value of the input signal and limiting.

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TABLE I

	<u>dm/dp</u>	<u>R/L</u>	<u>R/L in dB</u>	<u>% clipping</u>
	0	∞	∞	100
	1/16	23.3	27.4	94
5	1/8	12.0	21.6	89
	1/4	6.3	16.0	80
	1/2	3.5	10.9	67
	1	2.04	6.2	50
	2	1.29	2.2	33
10	4	.88	-1.1	20
	8	.64	-3.8	11
	16	.50	-6.0	6
	32	.40	-7.9	3

- In the above equations, the relationship, $R = G\sigma$, applies where G represents the gain prior to limiting and σ represents the root mean square speech signal level of the input signal. When the signal level σ changes, circuit 10 will adapt to a new state such that R/L or $G\sigma/L$ returns to the compression factor determined by dp and dm .
- 20 The initial rate of adaption is determined from the following equation:

$$dg/dt = f_c(dp(1-e^{-(\sqrt{2})L/(G\sigma)}) - dm(e^{-(\sqrt{2})L/(Gs)})) \quad (7)$$

- In equation (7), f_c represents the clock rate of clock 50. The path followed by the gain (G) is determined by solving
- 25 the following equations recursively:

$$dG = dp(1-e^{-(\sqrt{2})L/(G\sigma)}) - dm(e^{-(\sqrt{2})L/(Gs)}) \quad (8)$$

$$G = G + dG \quad (9)$$

Within equations (8) and (9), the attack and release times for circuit 10 are symmetric only for a

compression factor (R/L) of 2.04. The attack time corresponds to the reduction of gain in response to an increase in signal σ . Release time corresponds to the increase in gain after the signal level σ is reduced. For
5 a compression factor setting of 12, the release time is much shorter than the attack time. For a compression factor setting of .64 and .50, the attack time is much shorter than the release time. These latter values are preferable for a hearing aid.

10 As seen above, the rate of adaption depends on the magnitudes of dp and dm which are stored in registers 40 and 42. These 6-bit registers have a range from 1/128 dB to 63/128(dB). Therefore, at a sampling rate of 16kHz from clock 50, the maximum slope of the adaptive gain
15 function ranges from 125 dB/sec to 8000 dB/sec. For a step change of 32 dB, this corresponds to a typical range of time constant from 256 milliseconds to four milliseconds respectively. If dm is set to zero, the adaptive compression feature is disabled.

20 Fig. 2 discloses a circuit 60 which has a number of common circuit elements with circuit 10 of Fig. 1. Such common elements have similar functions and have been marked with common reference numbers. In addition to circuit 10, however, circuit 60 of Fig. 2 provides for a
25 programmable compression ratio. Circuit 60 has a gain control 66 which is connected to a register 62 by a line 64 and to gain register 24 by a line 68. Register 62 stores a compression factor. Gain control 66 takes the value stored in gain register 24 to the power of the
30 compression ratio stored in register 62 and outputs said power gain value via a line 70 to an amplifier 72. Amplifier 72 combines the power gain value on line 70 with the gain value stored in a register 74 to produce an
35 output gain on a line 76. An amplifier 78 receives the output gain via line 76 for controlling the gain of amplifier 78. Amplifier 78 amplifies the signal from

input 12 accordingly. The output signal from amplifier 78 is peak clipped by a limiter 80 and supplied as an output signal for circuit 60 at an output 82 in accordance with the invention.

5 To summarize the operation of circuit 60, the input to limiter 80 is generated by amplifier 78 whose gain is programmably set as a power of the gain setting stored in gain register 24, while the input to comparator 32 continues to be generated as shown in circuit 10 of
10 Fig. 1. Further, one of the many known functions other than the power function could be used for programmably setting the gain of amplifier 78.

The improvement in circuit 60 of Fig. 2 over circuit 10 of Fig. 1 is seen in Fig. 3 which shows the
15 input/output curves for compression ratios ranging from zero through two. The curve corresponding to a compression ratio of one is the single input/output curve provided by circuit 10 in Fig. 1. Circuit 60 of Fig. 2, however, is capable of producing all of the input/output curves shown
20 in Fig. 3.

In practice, circuit 10 of Fig. 1 or circuit 60 of Fig. 2 may be used in several parallel channels, each channel filtered to provide a different frequency response. Narrow band or broad band filters may be used
25 to provide maximum flexibility in fitting the hearing aid to the patient's hearing deficiency. Broad band filters are used if the patient prefers one hearing aid characteristic at low input signal levels and another characteristic at high input signal levels. Broad band
30 filters can also provide different spectral shaping depending on background noise level. The channels are preferably constructed in accordance with the filter/limit/filter structure disclosed in U.S. Patent No. 5,111,419 (hereinafter "the '419 patent") and incorporated
35 herein by reference.

Fig. 4 shows a 4-channel filter/limit/filter structure for circuit 10 of Fig. 1. While many types of filters can be used for the channel filters of Fig. 4 and the other Figs., FIR filters are the most desirable. Each of the filters F1, F2, F3 and F4 in Fig. 4 are symmetric FIR filters which are equal in length within each channel. This greatly reduces phase distortion in the channel output signals, even at band edges. The use of symmetric filters further requires only about one half as many registers to store the filter co-efficients for a channel, thus allowing a simpler circuit implementation and lower power consumption. Each channel response can be programmed to be a band pass filter which is contiguous with adjacent channels. In this mode, filters F1 through F4 have preset filter parameters for selectively passing input 12 over a predetermined range of audible frequencies while substantially attenuating any of input 12 not occurring in the predetermined range. Likewise, channel filters F1 through F4 can be programmed to be wide band to produce overlapping channels. In this mode, filters F1 through F4 have preset filter parameters for selectively altering input 12 over substantially all of the audible frequency range. Various combinations of these two cases are also possible. Since the filter coefficients are arbitrarily specified, in-band shaping is applied to the band-pass filters to achieve smoothly varying frequency gain functions across all four channels. An output 102 of a circuit 100 in Fig. 4 provides an adaptively compressed and filtered output signal comprising the sum of the filtered signals at outputs 30 in each of the four channels identified by filters F1 through F4.

Fig. 5 shows a four channel filter/limit/filter circuit 110 wherein each channel incorporates circuit 60 of Fig. 2. An output 112 in Fig. 5 provides a programmably compressed and filtered output signal comprising the sum of the filtered signals at outputs 82

in each of the four channels identified by filters F1 through F4.

The purpose of the adaptive gain factor in each channel of the circuitry of Figs. 4 and 5 is to maintain a specified constant level of envelope compression over a range of inputs. By using adaptive compressive gain, the input/output function for each channel is programmed to include a linear range for which the signal envelope is unchanged, a higher input range over which the signal envelope is compressed by a specified amount, and the highest input range over which envelope compression increases as the input level increases. This adaptive compressive gain feature adds an important degree of control over mapping a widely dynamic input signal into the reduced auditory range of the impaired ear.

The design of adaptive compressive gain circuitry for a hearing aid presents a number of considerations, such as the wide dynamic range, noise pattern and bandwidth found in naturally occurring sounds. Input sounds present at the microphone of a hearing aid vary from quiet sounds (around 30 dB SPL) to those of a quiet office area (around 50 dB SPL) to much more intense transient sounds that may reach 100 dB SPL or more. Sound levels for speech vary from a casual vocal effort of a talker at three feet distance (55 dB SPL) to that of a talker's own voice which is much closer to the microphone (80 dB SPL). Therefore, long term averages of speech levels present at the microphone vary by 25 dB or more depending on the talker, the distance to the talker, the orientation of the talker and other factors. Speech is also dynamic and varies over the short term. Phoneme intensities vary from those of vowels, which are the loudest sounds, to unvoiced fricatives, which are 12 dB or so less intense, to stops, which are another 18 dB or so less intense. This adds an additional 30 dB of dynamic range required for speaking. Including both long-term and

short-term variation, the overall dynamic range required for speech is about 55 dB. If a talker whispers or is at a distance much greater than three feet, then the dynamic range will be even greater.

5 Electronic circuit noise and processing noise limit the quietest sounds that can be processed. A conventional hearing aid microphone has an equivalent input noise figure of 25 dB SPL, which is close to the estimated 20 dB noise figure of a normal ear. If this
10 noise figure is used as a lower bound on the input dynamic range and 120 dB SPL is used as an upper bound, the input dynamic range of good hearing aid system is about 100 dB. Because the microphone will begin to saturate at 90 to 100 dB SPL, a lesser dynamic range of 75 dB is workable.

15 Signal bandwidth is another design consideration. Although it is possible to communicate over a system with a bandwidth of 3kHz or less and it has been determined that 3kHz carries most of the speech information, hearing aids with greater bandwidth result in
20 better articulation scores. Skinner, M.W. and Miller, J.D., Amplification Bandwidth and Intelligibility of Speech in Quiet and Noise for Listeners with Sensorineural Hearing Loss, 22:253-79 Audiology (1983). Accordingly, the embodiment disclosed in Fig. 1 has a 6 kHz upper
25 frequency cut-off.

 The filter structure is another design consideration. The filters must achieve a high degree of versatility in programming bandwidth and spectral shaping to accommodate a wide range of hearing impairments.
30 Further, it is desirable to use shorter filters to reduce circuit complexity and power consumption. It is also desirable to be able to increase filter gain for frequencies of reduced hearing sensitivity in order to improve signal audibility. However, studies have shown
35 that a balance must be maintained between gain at low frequencies and gain at high frequencies. It is

recommended that the gain difference across frequency should be no greater than 30 dB. Skinner, M.W., Hearing Aid Evaluation, Prentice Hall (1988). Further, psychometric functions often used to calculate a

5 "prescriptive" filter characteristic are generally smooth, slowly changing functions of frequency that do not require a high degree of frequency resolution to fit.

Within the above considerations, it is preferable to use FIR filters with transition bands of

10 1000 Hz and out of band rejection of 40 dB. The required filter length is determined from the equation:

$$L = ((-20\log_{10}(\sigma)-7.95) / (14.36TB/f_s)) + 1 \quad (10)$$

In equation (10), L represents the number of filter taps, σ represents the maximum error in achieving a target

15 filter characteristic, $-20 \log_{10}(\sigma)$ represents the out of band rejection in decibels, TB represents the transition band, and f_s is the sampling rate. See Kaiser, Nonrecursive Filter Design Using the I_0 -SINH Window Function, Proc., IEEE Int. Symposium on Circuits and

20 Systems (1974). For an out of band rejection figure of 35 dB with a transition band of 1000Hz and a sampling frequency of 16kHz, the filter must be approximately 31 taps long. If a lower out of band rejection of 30 dB is acceptable, the filter length is reduced to 25 taps. This

25 range of filter lengths is consistent with the modest filter structure and low power limitations of a hearing aid.

All of the circuits shown in Figs. 1 through 9 use log encoded data. See the '419 patent. Log encoding

30 is similar to u-law and A-law encoding used in Codecs and has the same advantages of extending the dynamic range, thereby making it possible to reduce the noise floor of the system as compared to linear encoding. Log encoding offers the additional advantage that arithmetic operations

are performed directly on the log encoded data. The log encoded data are represented in the hearing aid as a sign and magnitude as follows:

$$x = \text{sgn}(y)\log(|y|) / \log(B) \quad (11)$$

5 In equation (11), B represents the log base, which is positive and close to but less than unity, x represents the log value and y represents the equivalent linear value. A reciprocal relation for y as a function of x follows:

$$10 \quad y = \text{sgn}(x)B^{|x|} \quad (12)$$

If x is represented as sign and an 8-bit magnitude and the log base is 0.941, the range of y is ± 1 to $\pm 1.8 \times 10^{-7}$. This corresponds to a dynamic range of 134 dB. The general expression for dynamic range as a function of the
 15 log base B and the number of bits used to represent the log magnitude value N follows:

$$\text{dynamic range (dB)} = 20\log_{10}(B^{(2^N-1)}) \quad (13)$$

An advantage of log encoding over u-law encoding
 20 is that arithmetic operations are performed directly on the encoded signal without conversion to another form. The basic FIR filter equation, $y(n) = \sum a_i x(n-i)$, is implemented recursively as a succession of add and table lookup operations in the log domain. Multiplication is
 25 accomplished by adding the magnitude of the operands and determining the sign of the result. The sign of the result is a simple exclusive-or operation on the sign bits of the operands. Addition (and subtraction) are accomplished in the log domain by operations of
 30 subtraction, table lookup, and addition. Therefore, the sequence of operations required to form the partial sum of

products of the FIR filter in the log domain are addition, subtraction, table lookup, and addition.

Addition and subtraction in the log domain are implemented by using a table lookup approach with a
5 sparsely populated set of tables T_+ and T_- stored in a memory (not shown). Adding two values, x and y , is accomplished by taking the ratio of the smaller magnitude to the larger and adding the value from the log table T_+ to the smaller. Subtraction is similar and uses the log
10 table T_- . Since x and y are in log units, the ratio, $|y/x|$ (or $|x/y|$), which is used to access the table value, is obtained by subtracting $|x|$ from $|y|$ (or vice-versa). The choice of which of the tables, T_+ or T_- , to use is
15 determined by an exclusive-or operation on the sign bits of x and y . Whether the table value is added to x or to y is determined by subtracting $|x|$ from $|y|$ and testing the sign bit of the result.

Arithmetic roundoff errors in using log values for multiplication are not significant. With an 8-bit
20 representation, the log magnitude values are restricted to the range 0 to 255. Zero corresponds to the largest possible signal value and 255 to the smallest possible signal value. Log values less than zero cannot occur. Therefore, overflow can only occur for the smallest signal
25 values. Product log values greater than 255 are truncated to 255. This corresponds to a smallest signal value (255 LU's) that is 134 dB smaller than the maximum signal value. Therefore, if the system is scaled by setting the
30 the truncation errors of multiplication (255 LU) correspond to -134 dB relative to the maximum possible signal value (0 LU). In absolute terms, this provides a -4 dB SPL or -43 dB SPL spectrum level, which is well below the normal hearing threshold.

35 Roundoff errors of addition and subtraction are much more significant. For example, adding two numbers of

equal magnitude together results in a table lookup error of 2.4%. Conversely, adding two values that differ by three orders of magnitude results in an error of 0.1%. The two tables, T_+ and T_- , are sparsely populated. For a
5 log base of 0.941 and table values represented as an 8-bit magnitude, each table contains 57 nonzero values. If it is assumed that the errors are uniformly distributed (that each table value is used equally often on the average), then the overall average error associated with table
10 roundoff is 1.01% for T_+ and 1.02% for T_- .

Table errors are reduced by using a log base closer to unity and a greater number of bits to represent log magnitude. However, the size of the table grows and quickly becomes impractical to implement. A compromise
15 solution for reducing error is to increase the precision of the table entries without increasing the table size. The number of nonzero entries increases somewhat. Therefore, in implementing the table lookup in the digital processor, two additional bits of precision are added to
20 the table values. This is equivalent to using a temporary log base which is the fourth root of 0.941 (0.985) for calculating the FIR filter summation. The change in log base increases the number of nonzero entries in each of the tables by 22, but reduces the average error by a
25 factor of four. This increases the output SNR of a given filter by 12 dB. The T_+ and T_- tables are still sparsely populated and implemented efficiently in VLSI form.

In calculating the FIR equation, the table lookup operation is applied recursively $N-1$ times, where N
30 is the order of the filter. Therefore, the total error that results is greater than the average table roundoff error and a function of filter order. If it is assumed that the errors are uniformly distributed and that the input signal is white, the expression for signal to
35 roundoff noise ratio follows:

22

$$\frac{\varepsilon_y^2}{\sigma_y^2} = \frac{\varepsilon^2(c_1^2 + 2c_2^2 + \dots + (N-1)c_N^2)}{(c_1^2 + c_2^2 + \dots + c_N^2)} \quad (14)$$

In equation (14), ε_y^2 represents the noise variance at the output of the filter, σ_y^2 represents the signal variance at the output of the filter, and ε represents the average percent table error. Accordingly, the filter noise is dependent on the table lookup error, the magnitude of the filter coefficients, and the order of summation. The coefficient used first introduces an error that is multiplied by N-1. The coefficient used second introduces an error that is multiplied by N-2 and so on. Since the error is proportional to coefficient magnitude and order of summation, it is possible to minimize the overall error by ordering the smallest coefficients earliest in the calculation. Since the end tap values for symmetric filters are generally smaller than the center tap value, the error was further reduced by calculating partial sums using coefficients from the outside toward the inside.

In Figs. 4 and 5, FIR filters F1 through F4 represent channel filters which are divided into two cascaded parts. Limiters 26 and 80 are implemented as part of the log multiply operation. G_1 is a gain factor that, in the log domain, is subtracted from the samples at the output of the first FIR filter. If the sum of the magnitudes is less than zero (maximum signal value), it is clipped to zero. G_2 represents an attenuation factor that is added (in the log domain) to the clipped samples. G_2 is used to set the maximum output level of the channel.

Log quantizing noise is a constant percentage of signal level except for low input levels that are near the smallest quantizing steps of the encoder. Assuming a Laplacian signal distribution, the signal to quantizing noise ratio is given by the following equation:

$$\text{SNR(dB)} = 10\log_{10}(12) - 20 \log_{10} (|\ln(B)|) \quad (15)$$

For a log base of 0.941, the SNR is 35 dB. The quantizing noise is white and, since equation (15) represents the total noise energy over a bandwidth of 8kHz, the spectrum level is 39 dB less or 74 dB smaller than the signal level. The ear inherently masks the quantizing noise at this spectrum level. Schroeder, et al., Optimizing Digital Speech Coders by Exploiting Masking Properties of the Human Ear, Vol. 66(6) J.Acoust.Soc.Am. pp.1647-52 (Dec. 1979). Thus, log encoding is ideally suited for auditory signal processing. It provides a wide dynamic range that encompasses the range of levels of naturally occurring signals, provides sufficient SNR that is consistent with the limitation of the ear to resolve small signals in the presence of large signals, and provides a significant savings with regard to hardware.

The goal of the fitting system is to program the digital hearing aid to achieve a target real-ear gain. The real-ear gain is the difference between the real-ear-aided- response (REAR) and the real-ear-unaided-response (REUR) as measured with and without the hearing aid on the patient. It is assumed that the target gain is specified by the audiologist or calculated from one of a variety of prescriptive formulae chosen by the audiologist that is based on audiometric measures. There is not a general consensus about which prescription is best. However, prescriptive formulae are generally quite simple and easy to implement on a small host computer. Various prescriptive fitting methods are discussed in Chapter 6 of Skinner, M.W., Hearing Aid Evaluation, Prentice Hall (1988).

Assuming that a target real-ear gain has been specified, the following strategy is used to automatically fit the four channel digital hearing aid where each channel is programmed as a band pass filter which is contiguous with adjacent channels. The real-ear measurement system disclosed in U.S. Patent No. 4,548,082

(hereinafter "the '082 patent") and incorporated herein by reference is used. First, the patient's REUR is measured to determine the patient's normal, unoccluded ear canal resonance. Then the hearing aid is placed on the patient.

5 Second, the receiver and earmold are calibrated. This is done by setting G2 of each channel to maximum attenuation (-134dB) and turning on the noise generator of the adaptive feedback equalization circuit shown in the '082 patent. This drives the output of the hearing aid with a

10 flat-spectrum-level, pseudorandom noise sequence. The noise in the ear canal is then deconvolved with the pseudorandom sequence to obtain a measure of the output transfer characteristic (H_r) of the hearing aid. Third, the microphone is calibrated. This is done by setting the

15 channels to a flat nominal gain of 20 dB. The cross-correlation of the sound in the ear canal with the reference sound then represents the overall transfer characteristic of the hearing aid and includes the occlusion of sound by the earmold. The microphone

20 calibration (H_m) is computed by subtracting H_r from this measurement. Last, the channel gain functions are specified and filter coefficients are computed using a window design method. See Rabiner and Schafer, Digital Processing of Speech Signals, Prentice Hall (1978). The

25 coefficients are then downloaded in bit-serial order to the coefficient registers of the processor. The coefficient registers are connected together as a single serial shift register for the purpose of downloading and uploading values.

30 The channel gains are derived as follows. The acoustic gain for each channel of the hearing aid is given by:

$$\text{Gain} = H_m + H_r + H_n + G_{1n} + G_{2n} \quad (16)$$

The filter shape for each channel is determined by setting the Gain in equation (16) to the desired real-ear gain plus the open-ear resonance. Since G_{1n} and G_{2n} are gain constants for the channel and independent of frequency, they do not enter into the calculation at this point. The normalized filter characteristics is determined from the following equation.

$$H_n = 0.5 (\text{Desired Real-ear gain} + \text{open ear cal} - H_m - H_r + G_n) \quad (17)$$

H_m and H_r represent the microphone and receiver calibration measures, respectively, that were determined for the patient with the real ear measurement system and G_n represents a normalization gain factor for the filter that is included in the computation of G_{1n} and G_{2n} . H_m and H_r include the transducer transfer characteristics in addition to the frequency response of the amplifier and any signal conditioning filters. Once H_n is determined, the maximum output of each channel, which is limited by L , are represented by G_{2n} as follows:

$$G_{2n} = \text{MPO}_n - L - \text{avg}(H_n + H_r) - G_n \quad (18)$$

In equation (18), the "avg" operator gives the average of filter gain and receiver sensitivity at filter design frequencies within the channel. L represents a fixed level for all channels such that signals falling outside the range $\pm L$ are peak-clipped at $\pm L$. G_n represents the filter normalization gain, and MPO_n represents the target maximum power output. Overall gain is then established by setting G_{1n} as follows:

$$G_{1n} = 2G_n - G_{2n} \quad (19)$$

G_n represents the gain normalization factor of the filters that were designed to provide the desired linear gain for the channel.

By using the above approach, target gains typically are realized to within 3 dB over a frequency range of from 100 Hz to 6000 Hz. The error between the step-wise approximation to the MPO function and the target MPO function is also small and is minimized by choosing appropriate crossover frequencies for the four channels.

Because the channel filters are arbitrarily specified, an alternative fitting strategy is to prescribe different frequency-gain shapes for signals of different levels. By choosing appropriate limit levels in each channel, a transition from the characteristics of one channel to the characteristics of the next channel will occur automatically as a function of signal level. For example, a transparent or low-gain function is used for high-level signals and a higher-gain function is used for low-level signals. The adaptive gain feature in each channel provides a means for controlling the transition from one channel characteristic to the next. Because of recruitment and the way the impaired ear works, the gain functions are generally ordered from highest gain for soft sounds to the lowest gain for loud sounds. With respect to circuit 100 of Fig. 4, this is accomplished by setting G1 in gain register 22 very high for the channel with the highest gain for the soft sounds. The settings for G1 in gain registers 22 of the next succeeding channels are sequentially decreased, with the G1 setting being unity in the last channel which channel has the lowest gain for loud sounds. A similar strategy is used for circuit 110 of Fig. 5, except that G1 must be set in both gain registers 22 and 74. In this way, the channel gain settings in circuits 100 and 110 of Figs. 4 and 5 are sequentially modified from first to last as a function of the level of input 12.

The fitting method is similar to that described above for the four-channel fitting strategy. Real-ear measurements are used to calibrate the ear, receiver, and microphone. However, the filters are designed differently. One of the channels is set to the lowest gain function and highest ACG threshold. Another channel is set to a higher-gain function, which adds to the lower-gain function and dominates the spectral shaping at signal levels below a lower ACG threshold setting for that channel. The remaining two channels are set to provide further gain contributions at successively lower signal levels. Since the channel filters are symmetric and equal length, the gains will add in the linear sense. Two channels set to the same gain function will provide 6 dB more gain than either channel alone. Therefore, the channels filters are designed as follows:

$$H_1 = 1/2 D_1 \quad (20)$$

$$H_2 = 1/2 \log_{10} (10^{D_2} - 10^{D_1}) \quad (21)$$

$$H_3 = 1/2 \log_{10} (10^{D_3} - 10^{D_2} - 10^{D_1}) \quad (22)$$

$$H_4 = 1/2 \log_{10} (10^{D_4} - 10^{D_3} - 10^{D_2} - 10^{D_1}) \quad (23)$$

where: $D_1 < D_2 < D_3 < D_4$. D_n represents the filter design target in decibels that gives the desired insertion gain for the hearing aid and is derived from the desired gains specified by the audiologist and corrected for ear canal resonance and receiver and microphone calibrations as described previously for the four-channel fit. The factor, $1/2$, in the above expressions takes into account that each channel has two filters in cascade.

The processor described above has been implemented in custom VLSI form. When operated at 5 volts and at a 16-kHz sampling rate, it consumes 4.6mA. When operated at 3 volts and at the same sampling rate, it consumes 2.8 mA. When the circuit is implemented in a low-voltage form, it is expected to consume less than 1 mA

when operated from a hearing aid battery. The processor has been incorporated into a bench-top prototype version of the digital hearing aid. Results of fitting hearing-impaired subjects with this system suggest that prescriptive frequency gain functions are achieved within 3 dB accuracy at the same time that the desired MPO frequency function is achieved within 5 dB or so of accuracy.

For those applications that do not afford the computational resources required to implement the circuitry of Figs. 1 through 5, the simplified circuitry of Figs. 6 through 9 is used. In Fig. 6, a circuit 120 includes an input 12 which represents any conventional source of an input signal such as a microphone, signal processor, or the like. Input 12 also includes an analog to digital converter (not shown) for analog input signals if circuit 120 is implemented with digital components. Likewise, input 12 includes a digital to analog converter (not shown) for digital input signals if circuit 120 is implemented with analog components.

Input 12 is connected to a group of filters F1 through F4 and a filter S1 over a line 122. Filters F1 through F4 provide separate channels with filter parameters preset as described above for the multichannel circuits of Figs. 4 and 5. Each of filters F1, F2, F3 and F4 outputs an adaptively filtered signal via a line 124, 126, 128 and 130 which is amplified by a respective amplifier 132, 134, 136 and 138. Amplifiers 132 through 138 each provide a channel output signal which is combined by a line 140 to provide an adaptively filtered signal at an output 142 of circuit 120.

Filter S1 has parameters which are set to extract relevant signal characteristics present in the input signal. The output of filter S1 is received by an envelope detector 144 which detects said characteristics. Detector 144 preferably has a programmable time constant

for varying the relevant period of detection. When detector 144 is implemented in analog form, it includes a full wave rectifier and a resistor/capacitor circuit (not shown). The resistor, the capacitor, or both, are
5 variable for programming the time constant of detector 144. When detector 144 is implemented in digital form, it includes an exponentially shaped filter with a programmable time constant. In either event, the "on" time constant is shorter than the relatively long "off"
10 time constant to prevent excessively loud sounds from existing in the output signal for extended periods.

The output of detector 144 is a control signal which is transformed to log encoded data by a log transformer 146 using standard techniques and as more
15 fully described above. The log encoded data represents the extracted signal characteristics present in the signal at input 12. A memory 148 stores a table of signal characteristic values and related amplifier gain values in log form. Memory 148 receives the log encoded data from
20 log transformer 146 and, in response thereto, recalls a gain value for each of amplifiers 132, 134, 136 and 138 as a function of the log value produced by log transformer 146. Memory 148 outputs the gain values via a set of lines 150, 152, 154 and 156 to amplifiers 132, 134, 136
25 and 138 for setting the gains of the amplifiers as a function of the gain values. Arbitrary overall gain control functions and blending of signals from each signal processing channel are implemented by changing the entries in memory 148.

30 In use, circuit 120 of Fig. 6 may include a greater or lesser number of filtered channels than the four shown in Fig. 6. Further, circuit 120 may include additional filters, detectors and log transformers corresponding to filter S1, detector 144 and log
35 transformer 146 for providing additional input signal characteristics to memory 148. Still further, any or all

of the filtered signals in lines 124, 126, 128 or 130 could be used by a detector(s), such as detector 144, for detecting an input signal characteristic for use by memory 148.

5 Fig. 7 includes input 12 for supplying an input signal to a circuit 160. Input 12 is connected to a variable filter 162 and to a filter S1 via a line 164. Variable filter 162 provides an adaptively filtered signal which is amplified by an amplifier 166. A limiter 168
10 peak clips the adaptively filtered output signal of amplifier 166 to produce a limited output signal which is filtered by a variable filter 170. The adaptively filtered and clipped output signal of variable filter 170 is provided at output 171 of circuit 160.

15 Filter S1, a detector 144 and a log transformer 146 in Fig. 7 perform similar functions to the like numbered components found in Fig. 6. A memory 162 stores a table of signal characteristic values, related filter parameters, and related amplifier gain values in log form.
20 Memory 162 responds to the output from log transformer 146 by recalling filter parameters and an amplifier gain value as functions of the log value produced by log transformer 146. Memory 162 outputs the recalled filter parameters via a line 172 and the recalled gain value via a line 174.
25 Filters 162 and 170 receive said filter parameters via line 172 for setting the parameters of filters 162 and 170. Amplifier 166 receives said gain value via line 174 for setting the gain of amplifier 166. The filter coefficients are stored in memory 162 in sequential order
30 of input signal level to control the selection of filter coefficients as a function of input level. Filters 162 and 170 are preferably FIR filters of the same construction and length and are set to the same parameters by memory 162. In operation, the circuit 160 is also used
35 by taking the output signal from the output of amplifier 166 to achieve desirable results. Limiter 168 and

variable filter 170 are shown, however, to illustrate the filter/limit/filter structure disclosed in the '419 patent in combination with the pair of variable filters 162 and 170.

5 With a suitable choice of filter coefficients, a variety of level dependent filtering is achieved. When memory 162 is a random-access memory, the filter coefficients are tailored to the patient's hearing impairment and stored in the memory from a host computer
10 during the fitting session. The use of the host computer is more fully explained in the '082 patent.

 A two channel version of circuit 120 in Fig. 6 is shown in Fig. 8 as circuit 180. Like components of the circuits in Figs. 6 and 8 are identified with the same
15 reference numerals. A host computer (such as the host computer disclosed in the '082 patent) is used for calculating the F1 and F2 filter coefficients for various spectral shaping, for calculating entries in memory 148 for various gain functions and blending functions, and for
20 down-loading the values to the hearing aid.

 The gain function for each channel is shown in Fig. 9. A segment "a" of a curve G1 provides a "voice switch" characteristic at low signal levels. A segment "b" provides a linear gain characteristic with a spectral
25 characteristic determined by filter F1 in Fig. 8. A segment "c" and "d" provide a transition between the characteristics of filters F1 and F2. A segment "e" represents a linear gain characteristic with a spectral characteristic determined by filter F2. Lastly, segment
30 "f" corresponds to a region over which the level of output 142 is constant and independent of the level of input 12.

 The G1 and G2 functions are stored in a random access memory such as memory 148 in Fig. 8. The data stored in memory 148 is based on the specific hearing
35 impairment of the patient. The data is derived from an appropriate algorithm in the host computer and down-loaded

to the hearing aid model during the fitting session. The coefficients for filters F1 and F2 are derived from the patients residual hearing characteristic as follows: Filter F2, which determines the spectral shaping for loud sounds, is designed to match the patients UCL function. Filter F1, which determines the spectral shaping for softer sounds, is designed to match the patients MCL or threshold functions. One of a number of suitable filter design methods are used to compute the filter coefficient values that correspond to the desired spectral characteristic.

A Kaiser window filter design method is preferable for this application. Once the desired spectral shape is established, the filter coefficients are determined from the following equation:

$$C_n = \sum A_k (\cos(2\pi n f_k / f_s)) W_n \quad (24)$$

In equation (24), C_n represents the n 'th filter coefficient, A_k represents samples of the desired spectral shape at frequencies f_k , f_s represents the sampling frequency and W_n represents samples of the Kaiser Window. The spectral sample points, A_k , are spaced at frequencies, f_k , which are separated by the 6dB bandwidth of the window, W_n , so that a relatively smooth filter characteristic results that passes through each of the sample values.

The frequency resolution and maximum slope of the frequency response of the resulting filter is determined by the number of coefficients or length of the filter. In the implementation shown in Fig. 8, filters F1 and F2 have a length of 30 taps which, at a sampling rate of 12.5kHz, gives a frequency resolution of about 700 Hz and a maximum spectral slope of 0.04 dB/Hz.

Circuit 180 of Fig. 8 simplifies the fitting process. Through a suitable interactive display on a host computer (not shown), each spectral sample value A_k is


independently selected. While wearing a hearing aid which includes circuit 180 in a sound field, such as speech weighted noise at a given level, the patient adjusts each sample value A_k to a preferred setting for listening. The
5 patient also adjusts filter F2 to a preferred shape that is comfortable only for loud sounds.

Appendix A contains a program written for a Macintosh host computer for setting channel gain and limit values in a four channel contiguous band hearing aid. The
10 filter coefficients for the bands are read from a file stored on the disk in the Macintosh computer. An interactive graphics display is used to adjust the filter and gain values.

In view of the above, it will be seen that the
15 several objects of the invention are achieved and other advantageous results attained.

As various changes could be made in the above constructions without departing from the scope of the invention, it is intended that all matter contained in the
20 above description or shown in the accompanying drawings shall be interpreted as illustrative and not in a limiting sense.

Program WDHA

Wearable Digital Hearing Aid Control Program V. 1.0	
	Central Institute For The Deaf
	818 South Euclid Ave.
	St. Louis Mo. 63110
	Phone: 314-652-3200
Supported in part by:	
The Rehabilitation Research And Development Service	
Dept. of Medicine and Surgery: Veterans Administration	

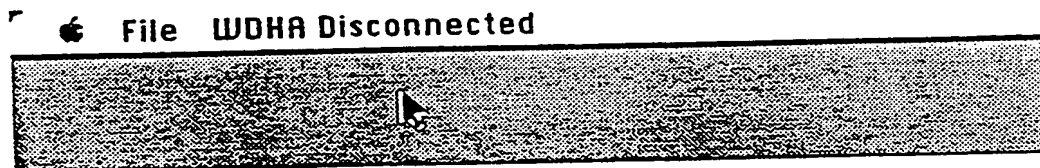
General Overview

A program entitled "WDHA" has been written for the Macintosh personal computer. When a wearable digital hearing aid is attached to the Macintosh's SCSI bus peripheral interface, the user of the WDHA program can alter the operation of the hearing aid via an easy to use Macintosh style user interface.

Using the WDHA Program

Starting The Program

Upon starting the program, the Macintosh interrogates the hearing aid to determine which program it is running. If the hearing aid responds appropriately, a menu containing the options which apply to that particular program appears in the menu bar. If no response is received from the hearing aid, the menu entitled "WDHA Disconnected" appears in the menu bar, as follows:

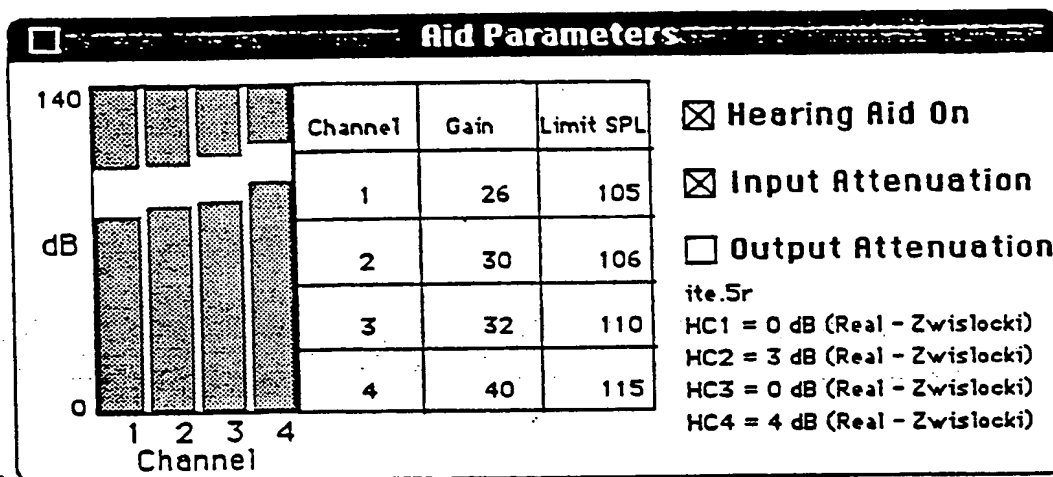


Should this menu appear, this indicates that there is some problem with the hearing aid. The source of this problem could be that the hearing aid is truly disconnected, that it is simply turned off, or that the hearing aid battery is dead. Upon correcting the problem,

choose the "New WDHA Program" menu entry to activate the proper menu for the hearing aid.

The Aid Parameters Window

The four channel hearing aid programs have the titles Aid12 through Aid14. Choosing the "Aid Parameters" menu entry will cause the aid parameters window to be displayed, as follows:



The bar graph and chart depict the current settings of the gains and limits for each channel of the hearing aid. A gain or limit setting can be changed by dragging the appropriate bar up or down with the mouse. The selected bar will blink when it is activated, and can be moved until the mouse is released, at which point the hearing aid is updated with the new values.

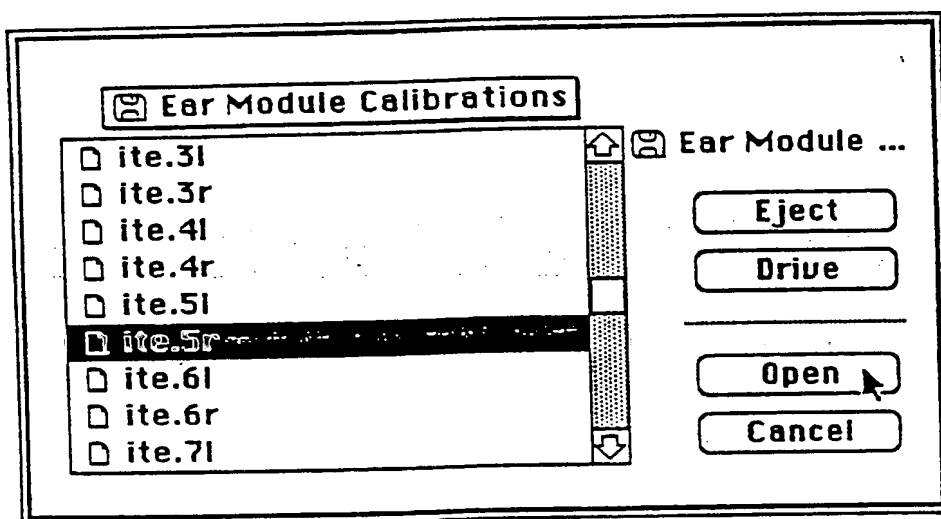
The control buttons indicate whether the hearing aid is on or off (i.e. whether the hearing aid program is running), and whether the input or output attenuators are switched on or off. Any of these settings can be changed simply by clicking on the appropriate buttons.

Ear Module Calibration

The File menu has an option called "Calibrate Ear Module" which should be used whenever the program is started or an ear module is inserted (or re-inserted) in a patient's ear. Proper use of

this option insures that the gains actually generated by the hearing aid are as close to the gains indicated by the program as possible.

The lower right hand corner of the Aid Parameters window displays the results of the most recent ear module calibration, including the name of the calibration file and the four Hc values, where Hc is the difference between the real ear pressure measured in the ear canal and the standard pressure measured on a Zwislocki at the center frequency of each channel. After choosing this option the user must open the file containing the ear module coefficients, by double clicking on the file's name, via a standard Macintosh dialog box:



The program will then play a series of four tones in the patient's ear, using the power measurement to determine the real pressure in the ear canal.

The file containing the ear module coefficients should be created with a text editor and saved as a text-only file. The file contains all the H values for a given ear module, separated by tabs, spaces, or carriage returns. It should begin with the four He values, followed by the Hr values, then Hc, and then Hp. The values entered for the Hc values can be arbitrary, since the program calculates them and stores them into the file. An ear module file as you would enter it might look as follows:

```
-100 -85 -90 -84 121 116 127 120
0
0
```

```

0
0
-124 -121 -134 -143

```

Here the first row contains both the four H_e values and the four H_r values. Following this are four zeros (since the H_c values are unknown). The sixth row contains the H_p values. Note that values are arbitrarily separated by tabs, spaces, or carriage returns.

After doing an ear module calibration with the program, the new H_c values are displayed in the Aid Settings window, and also written to the same file, with the data re-formatted into a separate row for each H value, as follows:

```

-100 -85 -90 -84
121 116 127 120
-5 -4 -10 0
-124 -121 -134 -143

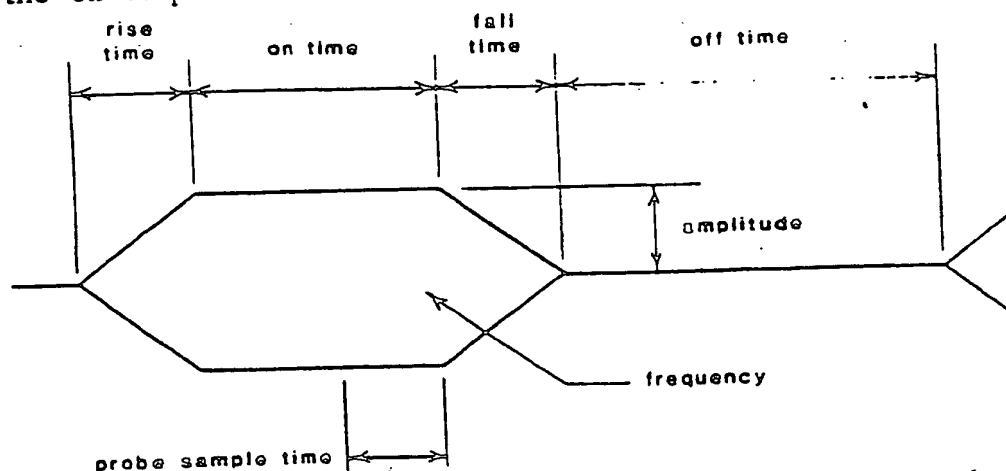
```

The Tone Parameters Window

The four channel programs also have the ability to play pure tones for audiometric purposes. The Tone Parameters window is available to activate these functions. Choosing the "Tone Parameters" menu entry will cause the Tone Parameters window to be displayed, as follows:

Tone Parameters		
Tone burst count?	3	<input checked="" type="checkbox"/> Hearing Aid On
Rise time sample count?	309	<input type="checkbox"/> Input Attenuation
Signal on sample count?	2455	<input type="checkbox"/> Output Attenuation
Fall time sample count?	309	<input type="radio"/> Field Mike
Signal off sample count?	3069	<input checked="" type="radio"/> Probe Mike
Frequency?	2000	<input type="button" value="Start"/>
Atten re max out (dB)?	20	
Power = -12.816046		

The text boxes specify the number of tone bursts to generate and the envelope of the tone bursts generated, as follows:



All times are specified in number of sample periods, and cannot exceed 32767 sample periods. The test is initiated by clicking on the start button. The control buttons act just as in the aid parameters window.

Loading Filter Taps

The programs titled Aid13 and Aid14 have the capability to download filter tap coefficients to the hearing aid. The coefficients are read into memory from a text file which the user creates with any standard text editor. The coefficients in these files are signed integers such as "797" or "-174" (optionally be followed by a divisor, such as in "-12028/2") and must be separated by spaces, tabs, or carriage returns.

The Aid13 program has 32 taps per filter, and the Aid14 program has 31 taps per filter, but since the filters are symmetric about the center tap you only provide half this number of taps, or 16 taps per filter. Thus the files contain 64 coefficients for the 4 channels. For example, the file titled TapsFour has the following format:

```
-535/4 -431/4 -254/4 0 333/4 743/4 1220/4 1750/4
2315/4 2892/4 3545/4 3977/4 4432/4 4797/4 5052/4 5183/4
-34/2 -231/2 -223/2 0 292/2 398/2 77/2 -745/2
-1873/2 -2869/2 -3212/2 -2535/2 -831/2 1483/2 3683/2 5021/2
-83/2 502/2 859/2 0 -1128/2 -866/2 189/2 128/2
-442/2 890/2 3076/2 1605/2 -3814/2 -6280/2 -922/2 6543/2
```


528/2 -167/2 -446/2 0 585/2 288/2 -1203/2 242/2
 442/2 1525/2 -2946/2 797/2 -174/2 6280/2 -12028/2 6482/2

The option to download coefficients is enabled by choosing the "Tap Filter Load" menu entries. The Macintosh will then present the standard open file dialog box, which you use to specify the name of the appropriate text file.

Program Design

The program is written in 68000 Assembly Language using the Macintosh Development System assembler, from Apple.

The program has been structured into separate managers for each of the program's functions. A separate file contains the functions associated with each manager. For example, the Parameter Settings (or "PS") manager is contained in the file WDHAPS.Asm, and includes all routines associated with the Aid Parameters window.

Below is a description of each manager, its function, and the routines contained in each.

WDHA.Asm

The overall program structure is typical of a Macintosh application in that it has an event loop which dequeues events from the event queue, and then branches to code which processes each particular type of event. WDHA.Asm contains the WDHA program's event loop.

WDHAPS.Asm

The Parameter Settings ("PS") manager contains all routines associated with the Aid Parameters window, which allows the user to control the gains and limits of each of the channels in the four channel programs. Specifically, these routines are as follows:

WDHAPSOpen - Create and display the Aid Parameters window.

WDHAPSClose - Close the Aid Parameters window and dispose the memory associated with it.

WDHAPSShow - Make the Aid Parameters window visible.

WDHAPSHide - Make the Aid Parameters window invisible.

WDHAPSDraw - Update the contents of the Aid Parameters window.

WDHAPSControl - Cause the appropriate modification of the Aid Parameters window when a mousedown event occurs within it's content region.

WDHAPSIS - Given a window pointer, this routine determines if it is the Aid Parameters window or not.

WDHAPSSetParam - Update the hearing aid to contain the settings specified in the Aid Parameters window.

WDHATC.Asm

The TC manager contains all routines associated with the Tone Parameters window, which allows the user to specify the parameters for the test/calibrate function of the four channel program, and initiate the test. Specifically, these routines are as follows:

WDHATCOpen - Create and display the Tone Parameters window.

WDHATCClose - Close the Tone Parameters window and dispose the memory associated with it.

WDHATCShow - Make the Tone Parameters window visible.

WDHATCHide - Make the Tone Parameters window invisible.

WDHATCDraw - Update the contents of the Tone Parameters window.

WDHATCControl - Cause the appropriate modification of the Tone Parameters window when a mousedown event occurs within it's content region.

WDHATCIS - Given a window pointer, this routine determines if it is the Tone Parameters window or not.

WDHATCIdle - Blink the text caret of the Tone Parameters window.

WDHATCKey - Insert a key press into the active text box of the Tone Parameters window.

WDHATCDoTest - Initiate a test by the hearing aid program, using the parameters specified by the Tone Parameters window.

EarModuleCalibrate - Compute the Hc values for each of the four channels (this routine uses the test/calibrate function of the hearing aid to figure the real ear pressure at the center frequency of each channel).

WDHASCSI.Asm

The SCSI manager contains all routines which send record structures to the hearing aid via the SCSI bus.

- SetParam - Send the four channel parameter record (containing the gains and limits) to the four channel hearing aid program.
- SetCoefficients - Send out the filter tap coefficients to the four channel hearing aid program.
- SetFileParams - Send the parameters required by the spectral shaping program.
- wdhatest - Initiate a pure tone test by sending the test/calibrate record to the hearing aid.

WDHAFC.Asm

The WDHA program accesses some numerical values it needs by reading them in from text files. The File Coefficients (FC) manager contains routines which access these text files.

WDHAFCSet - This routine is called when the user selects the "Load Filter Taps" menu option. It uses the SFGetFile dialog to get the name of a text file containing filter coefficients, convert the contents to integer form, and then downloads them to the hearing aid.

WDHASetFileParams - This routine is used to download parameters to the Spectral Shaping hearing aid program. It uses the SFGetFile dialog to get the name of a text file containing the spectral shaping parameters, converts the contents to integer form, then downloads them to the hearing aid.

WDHACalEarModFile - This routine is called when the user calibrates the ear module. It uses the SFGetFile dialog to get the name of a text file containing ear module H Tables, and converts it's contents to integer form in memory. Then it calibrates the ear module using the TC manager function EarModuleCalibrate. Finally, it writes the new H Tables over the same file.

WDHAMenu.Asm

The Menu manager contains all routines associated with the WDHA program's menu bar.

MakeMenus - Create the Menu bar containing the accessory, file, and hearing aid menus, and display it on the screen.

MenuBar - When the main event loop gets a mouseDown event located in the menu Bar, this routine calls the appropriate code to handle the selection.

SetProgMenu - This routine interrogates the hearing aid to determine which program it is currently running, then places the appropriate menu in the menu bar.

Programmer's Note -

As explained earlier, the WDHA program has separate pulldown menus defined for each program which runs on the hearing aid, giving the options available for that particular program. It is not difficult to add a new menu to the hearing aid program. The following example shows the steps one would follow to add a new aid menu (in this case 'Aid17') to the menu bar.

First of all, the constants needed for the menu must be defined with equate statements. You must define the code returned by the aid program when it is interrogated by the Macintosh, the identifier for the menu itself (as required by the NewMenu toolbox function), and the offset within the menu handles declarations where this handle will reside (the handles are defined in a sequential block of memory near the end of the Menu.Asm file).

```
Aid17ID    equ  -17    ; aid program id returned by interrogating the
aid.
Aid17Menu   equ  17     ; Unique menu identifier
menuaid17equ 40      ; 10*4=menuhandle offset (this is the tenth
handle)
```

Next you would declare the location to store the menu's handle at the end of the menu handles declarations:

```
dc.l 0      ; Aid17 menu handle
```

Next one would add code to the MakeMenus routine to create the new menu (simply cut and paste the code which creates one of the current menus and modify it accordingly).

You would also modify the SetProgMenu routine to handle the new menu (once again simply replicate the code sections which handle one of the old menus, and change the menu names appropriately).

Finally, you would modify the MenuBar routine to handle your new menu. If all the options contained in your menu are also in the

other hearing aid menus, you can call the InAidMenu procedure (as the other menus do), otherwise you must define your own procedure to call.

WDHADisk.Asm

The disk manager contains routines used to access disk files on the Macintosh.

DiskCreate - Create a new file.

DiskRead - Read sectors from a file.

DiskWrite - Write sectors to a file.

DiskEject - Eject a disk.

DiskOpen - Open a file.

DiskClose - Close a file

DiskSetFPos - Set the position of a file's read/write mark.

DiskSetEOF - Set the location of the end of file marker for a file.

DiskSetFInfo - Set the finder information for a file.

```

Include MacTraps.D
Include ToolEquX.D
Include SysEquX.D
Include QuickEquX.D
Include MDS2:WDHAPS.hdr
Include MDS2:WDHATC.hdr
Include MDS2:WDHAMenu.hdr

```

```

;-----
; WDMA program
;   This program controls several Macintosh windows which allow the user to
;   manipulate the digital hearing aid. The Macintosh communicates with the aid
;   by sending records via the SCSI port.
;   This particular file is a "standard" Macintosh style event loop
;   which dequeues each event and calls the appropriate routine to handle the event.
;   Additional files contain routines associated with each control window.
;   Executing the program should provide an overall understanding of the function
;   of these windows. Specifically, the packages used are:

```

```

;   The WDMA Paramater Settings Window Manager - in WDHAPS.Asm
;   The WDMA Test/Calibrate Window Manager - in WDHATC.Asm

```

```

;   In addition, the following files contain various utility routines:

```

```

;   WDHAMenu.Asm - sets up the menus
;   WDHASCSI.Asm - low level routines for communicating through the SCSI bus.
;   WDHAFSC.Asm - contains high-level routines for downloading coefficient
;   files to the hearing aid.
;   WDHADisk.Asm - routines for doing disk access.

```

```

;-----External Definitions-----

```

```

XDEF Start
XDEF EventLoop
XDEF Update
XDEF What
XDEF When
XDEF EventRecord
XDEF WWindow
XDEF Message
XDEF Where
XDEF Modify

```

```

;----- Constant Definitions -----

```

```

ActiveBit equ 0 ;Bit position of de/activate in modify

```

```

;----- Code Starts Here -----

```

```

Start:
    bsr InitManagers ; Initialize ToolBox
    bsr WDHAPSOOpen ; Create the parameter settings window.
    bsr WDHAPSHide ; Don't leave it open though.
    bsr WDHATCOOpen ; Create the test/calibrate window.
    bsr WDHATCHide ; Don't leave it open though.

```

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```

        bsr      MakeMenus          ; Set up the menus
EventLoop:
        _SystemTask                ; Give System some time
        bsr      WDHATCIdle         ; Blink the test window's caret
; FUNCTION      GetNextEvent(eventMask: INTEGER;
;                  VAR theEvent: EventRecord) : BOOLEAN
        CLR      -(SP)              ; Clear space for result
        MOVE     #$0FFF, -(SP)      ; Allow 12 low events
        PEA      EventRecord        ; Place to return results
        _GetNextEvent              ; Look for an event
        MOVE     (SP)+, D0          ; Get result code
        BEQ      EventLoop          ; No event... Keep waiting
        BSR      HandleEvent        ; Go handle event
        bra      EventLoop          ; return to eventloop call

```

HandleEvent:

```

; Use the event number as an index into the Event table. These 12 events
; are all the things that could spontaneously happen while the program is
; in the main loop.

```

```

        MOVE     What, D0            ; Get event number
        ADD      D0, D0              ; *2 for table index
        MOVE     EventTable(D0), D0 ; Point to routine offset
        JMP      EventTable(D0)     ; and jump to it

```

EventTable:

```

DC.W      OtherEvent-EventTable ; Null Event (Not used)
DC.W      MouseDown-EventTable ; Mouse Down
DC.W      OtherEvent-EventTable ; Mouse Up (Not used)
DC.W      KeyEvent-EventTable  ; Key Down
DC.W      OtherEvent-EventTable ; Key Up (Not used)
DC.W      KeyEvent-EventTable  ; Auto Key
DC.W      UpDate-EventTable    ; Update
DC.W      OtherEvent-EventTable ; Disk (Not used)
DC.W      Activate-EventTable  ; Activate
DC.W      OtherEvent-EventTable ; Abort (Not used)
DC.W      OtherEvent-EventTable ; Network (Not used)
DC.W      OtherEvent-EventTable ; I/O Driver (Not used)

```

----- Event Actions -----

OtherEvent:

rts

Activate:

```

; An activate event is posted by the system when a window needs to be
; activated or deactivated. The information that indicates which window
; needs to be updated was returned by the NextEvent call.

```

```

        btst     #ActiveBit, Modify ; Activate?
        beq      Deactivate         ; No, go do Deactivate
; Bring it to the front
        move.l    Message, -(sp)

```

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```

        _BringToFront
; Show it
        move.l      Message,-(sp)
        _ShowWindow
; Select it
        move.l      Message,-(sp)
        _SelectWindow
        rts

```

```

Deactivate:
        rts

```

```

Update:
; The window needs to be redrawn.

```

```

;PROCEDURE  BeginUpdate (theWindow: WindowPtr);
        MOVE.L      message,-(SP)          ; Get pointer to window
        _BeginUpDate      ; Begin the update
        move.l      message,-(sp)
        bsr         WDHATCIS              ; Was it our TC window?
        tst.w       (sp)+
        BEQ         DontTCDraw
        bsr         WDHATCDraw            ; Draw the TC window.
        bra         DoneDraw
DontTCDraw:
        move.l      message,-(sp)
        bsr         WDHAPCIS              ; Was it our PS window?
        tst.w       (sp)+
        BEQ         DontPSDraw
        bsr         WDHAPSDraw            ; Draw the PS window.
        bra         DoneDraw
DontPSDraw:
DoneDraw:

```

```

;PROCEDURE  EndUpdate (theWindow: WindowPtr);
        MOVE.L      message,-(SP)          ; Get pointer to window
        _EndUpdate      ; and end the update
        rts

```

```

MouseDown:
; If the mouse button was pressed, we must determine where the click
; occurred before we can do anything. Call FindWindow to determine
; where the click was; dispatch the event according to the result.

```

```

;FUNCTION   FindWindow (thePt: Point;
;                     VAR whichWindow: WindowPtr); INTEGER;
;
        CLR         -(SP)                  ; Space for result
        MOVE.L      Where,-(SP)            ; Get mouse coordinates
        PEA         WWindow                ; Event Window
        _FindWindow      ; Who's got the click?
        MOVE        (SP)+,D0               ; Get region number
        ADD         D0,D0                  ; *2 for index-into table
        MOVE        WindowTable(D0),D0    ; Point to routine offset

```



```

        JMP          WindowTable(D0)          ; Jump to routine

WindowTable:
        DC.W        other-WindowTable        ; In Desk (Not used)
        DC.W        MenuBar-WindowTable      ; In Menu Bar
        DC.W        SystemEvent-WindowTable  ; System Window (Not used)
        DC.W        Content-WindowTable      ; In Content
        DC.W        Drag-WindowTable         ; In Drag
        DC.W        Grow-WindowTable         ; In Grow
        DC.W        GoAway-WindowTable       ; In Go Away

Other:
        rts

SystemEvent:
; Call SystemClick to handle the desk accessory windows.
        pea         EventRecord
        move.l      wwindow,-(sp)
        _SystemClick
        rts

Content:
; Was it in the content of an active window?
        clr.l       -(sp)
        _FrontWindow
        move.l      (sp)+,d1                  ; Get the FrontWindow in d1
        cmp.l       wwindow,d1              ; Are they the same?
        beq         WasActive
        move.l      wwindow,-(sp)           ; It wasn't
        _SelectWindow                          ; So select it.
        bra        DoneContent

WasActive:
        move.l      wwindow,-(sp)
        bsr         WDHAPSIS                ; Was it our PS window?
        tst.w       (sp)+
        beq         NotPSContent
        move.l      where,-(sp)
        bsr         WDHAPSCControl          ; Handle the event.
        bra        DoneContent

NotPSContent:
        move.l      wwindow,-(sp)
        bsr         WDHATCIS                ; Was it our TC window?
        tst.w       (sp)+
        beq         NotTCCContent
        move.l      where,-(sp)
        bsr         WDHATCCControl          ; Handle the event
        bra        DoneContent

NotTCCContent:
DoneContent:
        rts

Drag:
; The click was in the drag bar of the window. Draggit.
; DragWindow (theWindow:WindowPtr; startPt: Point; boundsRect: Rect);

```

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```

MOVE.L wwindow,-(SP);Pass window pointer
MOVE.L where,-(SP) ;mouse coordinates
PEA bound ;and boundaries
_DragWindow ;Drag Window
rts

```

```

Grow:
; The click was in the grow box
NoGrow: rts

```

```

GoAway: ; Close the Window
        clr.b -(sp) ; make room for a Boolean
        move.l wwindow,-(sp)
        move.l where,-(sp)
        _TrackGoAway ; Track It
        tst.b (sp)+ ; Did they stay in the box?
        beq NoGoAway ; If no then don't close.

```

```

JustHide:
;PROCEDURE HideWindow (theWindow: WindowPtr)
        MOVE.L wwindow,-(SP) ; Pass window pointer
        _HideWindow ; Hide the Window
NoGoAway:
rts

```

```

KeyEvent:
        CLR.L -(SP) ; Space for result
        _FrontWindow ; Get window pointer on stack
        bsr WDHATCIS ; Was it our TC window?
        tst.w (sp)+
        beq TCNotActive
        move.w message+2,-(sp) ; get the char
        bsr WDHATCKey ; Insert it in the active text box
TCNotActive:
rts

```

```

; InitManagers initializes all the ToolBox managers. You should call
; InitManagers once at the beginning of your program if you are using
; any of the ToolBox routines.

```

```

InitManagers:
        pea -4(a5)
        _InitGraf
        _InitFonts
        move.l #$0000FFFF,d0
        _FlushEvents
        _InitWindows
        _InitMenus
        clr.l -(sp)
        _InitDialogs
        _TEInit
        _InitCursor
        rts

```

; WDHA header file
; this file must be included to access the data structures contained in
; the file WDHA.Asm

XREF EventLoop
XREF Update
XREF EventRecord
XREF What
XREF Message
XREF When
XREF Where
XREF Modify
XREF WWindow

TRUE EQU 1
FALSE EQU 0

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```
;WDHAMac.txt
;macros for WDMA program
;12/27/86 AME
```

```
;Dialog
;Macro
```

```
Macro Dialog xpos,ypos,txtstring,result =
move.w{xpos},-(SP)
move.w{ypos},-(SP)
_MoveTo
pea '{txtstring}'
_DrawString
pea KeyBuf
bsr GetStr

lea keybuf,a0
move.w#1,-(SP)
_Pack7 ;StringToNum
move.wd0,{result}
|
```

```
;DispString
;Macro
```

```
Macro DispString xpos,ypos,txtstring =
move.w{xpos},-(SP)
move.w{ypos},-(SP)
_MoveTo
pea '{txtstring}'
_DrawString
|
```

```
;DispValue
;Macro
```

```
Macro DispValue xpos,ypos,label,value =
movem.l a0-a6/d0-d7, -(sp)
move.w{xpos},-(SP)
move.w{ypos},-(SP)
_MoveTo
pea '{label}'
_DrawString

lea KeyBuf,a0
move.l {value},d0
move.w#0, -(SP) ;Select NumToString
_Pack7

pea KeyBuf
_DrawString
movem.l (sp)+,a0-a6/d0-d7
|
```

```
;DispWValue
;Macro
```

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```
Macro DispWValue xpos,ypos,label,value =  
    movem.l      a0-a6/d0-d7,-(sp)  
    move.w(xpos),-(SP)  
    move.w(ypos),-(SP)  
    _MoveTo  
    pea    '(label)'  
    _DrawString  
  
    lea    KeyBuf,a0  
    move.w(value),d0  
    ext.l  d0  
    move.w#0,-(SP)          ;Select NumToString  
    _Pack7  
  
    pea    KeyBuf  
    _DrawString  
    movem.l      (sp)+,a0-a6/d0-d7  
    |
```

```
; WDHAMenu.Asm
; This file contains routines which create and manipulate the menus used in
; the WDHA program.
```

```
Include MacTraps.D
Include ToolEquX.D
Include SysEquX.D
Include QuickEquX.D
Include MDS2:WDHAMac.txt
Include MDS2:WDHA.hdr
Include MDS2:WDHAPS.hdr
Include MDS2:WDHATC.hdr
Include MDS2:WDHAFC.hdr
Include MDS2:WDHASCSI.hdr
```

```
xdef    MakeMenus
xdef    MenuHandles
xdef    MenuBar
```

```
AppleMenu    EQU    1
AboutItem    EQU    1
menuapple     equ 0   ;menuhandle offset
```

```
FileMenu      EQU    2
QuitItem      EQU    1
menufile      equ 4   ;menuhandle offset
```

```
; Now the aid menus. All have a 'new program' entry, and a blank line.
NewProgItem   EQU    1
AidBlank      EQU    2
```

```
Aid12ID       EQU    -12   ; program version id
Aid12Menu     EQU    5
SetItem       EQU    3
TestItem      EQU    4
menuaid12     equ 8   ;menuhandle offset
```

```
Aid13ID       EQU    -13   ; program version id
Aid13Menu     EQU    6
FCItem        EQU    5
menuaid13     equ 12  ;menuhandle offset
```

```
Aid14ID       EQU    -14   ; program version id
Aid14Menu     EQU    7
menuaid14     equ 16  ;menuhandle offset
```

```
SS15ID        EQU    -100
SS15Menu      EQU    8
LoadItem      EQU    3
menuss15      equ    20
```

```
NoneMenu      EQU    9
menunone      equ    24
```

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```

; Name: MakeMenus
; Function: MakeMenus creates and displays the menu bar.
; Input: None
; Output: None
MakeMenus:
; Clear menu bar
    _ClearMenuBar

    lea    MenuHandles,a4
; First add Apple Menu
; Make it.
    clr.l    -(sp)                ;space for function result
    move.w #AppleMenu,-(sp)        ;first menu
    pea     AppleName              ;apple character
    _NewMenu
    move.l (sp)+,menuapple(a4)     ;store handle
; Add entries
    move.l menuapple(a4),-(sp)     ;push handle again
    pea     'About WDHA;(-'        ;push menu item
    _AppendMenu
    move.l menuapple(a4),-(sp)     ;push handle again
    move.l #'DRVR',-(sp)           ;load all drivers
    _AddResMenu
; Insert it in the menu bar.
    move.l menuapple(a4),-(sp)     ;push handle again
    move.w #0,-(sp)                ;insert at end
    _InsertMenu

; Now add File Menu
; Make it.
    clr.l    -(sp)                ;space for function result
    move.w #FileMenu,-(sp)         ;second menu
    pea     'File'                 ;menu title
    _NewMenu
    move.l (sp)+,menufile(a4)      ;store handle
; Add entries
    move.l menufile(a4),-(sp)      ;push handle again
    pea     'Quit'                 ;push menu item
    _AppendMenu
; Insert it in the menu bar.
    move.l menufile(a4),-(sp)      ;push handle again
    move.w #0,-(sp)                ;insert at end
    _InsertMenu

; Now create the WDHA program menus.
; none
    clr.l    -(sp)                ;space for function result
    move.w #NoneMenu,-(sp)         ;second menu
    pea     'WDHA Disconnected'    ;menu title
    _NewMenu
    move.l (sp)+,menunone(a4)      ;store handle
; Add entries.
    move.l menunone(a4),-(sp)      ;push handle
    pea     'New WDHA Program;(-' ;menu items.

```

_AppendMenu

```

; aid12
    clr.l    -(sp)                ;space for function result
    move.w#Aid12Menu,-(sp)
    pea 'Aid12'                  ;menu title
    _NewMenu
    move.l (sp)+,menuaid12(a4)    ;store handle
;Add entries.
    move.l menuaid12(a4),-(sp)    ;push handle
    pea 'New WDHA Program;(-;4 Channel Parameters;Test Calibrate' ;menu items.
    _AppendMenu

; aid13
    clr.l    -(sp)                ;space for function result
    move.w#Aid13Menu,-(sp)
    pea 'Aid13'                  ;menu title
    _NewMenu
    move.l (sp)+,menuaid13(a4)    ;store handle
;Add entries.
    move.l menuaid13(a4),-(sp)    ;push handle
    pea 'New WDHA Program;(-;4 Channel Parameters;Test Calibrate;32 Tap Filter Load'
;menu items.
    _AppendMenu

; aid14
    clr.l    -(sp)                ;space for function result
    move.w#Aid14Menu,-(sp)
    pea 'Aid14'                  ;menu title
    _NewMenu
    move.l (sp)+,menuaid14(a4)    ;store handle
;Add entries.
    move.l menuaid14(a4),-(sp)    ;push handle
    pea 'New WDHA Program;(-;4 Channel Parameters;Test Calibrate;31 Tap Filter Load'
;menu items.
    _AppendMenu

; SS15
    clr.l    -(sp)                ;space for function result
    move.w#SS15Menu,-(sp)
    pea 'SS15'                  ;menu title
    _NewMenu
    move.l (sp)+,menuss15(a4)     ;store handle
;Add entries.
    move.l menuss15(a4),-(sp)     ;push handle
    pea 'New WDHA Program;(-;Parameter Load' ;menu items.
    _AppendMenu

;Insert one in the menu bar since SetProgMenu deletes one.
    move.l menunone(a4),-(sp)    ;push handle again
    move.w#0,-(sp)               ;insert at end
    _InsertMenu

; Set the proper WDHA program menu

```


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```

        bsr      SetProgMenu
        rts

; Name: SetProgMenu
; Function: This routine interrogates the hearing aid to determine which
;           program it is currently running, then places the appropriate menu
;           in the menu bar.
; Input: None
; Output: None
SetProgMenu:
; Close windows so that no inappropriate windows remain.
        bsr      WDHAPSHide
        bsr      WDHATCHide
; Delete the old menu (whichever it is)
        move.w #Aid12Menu, -(sp)
        _DeleteMenu
        move.w #Aid13Menu, -(sp)
        _DeleteMenu
        move.w #Aid14Menu, -(sp)
        _DeleteMenu
        move.w #SS15Menu, -(sp)
        _DeleteMenu
        move.w #NoneMenu, -(sp)
        _DeleteMenu
; Default to NoneMenu
        lea       MenuHandles, a4
        move.l menunone(a4), -(sp)
        move.w #0, -(sp)
        _InsertMenu
; redraw the bar
        _DrawMenuBar
        move.w #0, -(sp)                ;clear any highlighting.
        _HiLiteMenu
; Now check what it is
        clr.w  -(sp)
        bsr    SCSIInterrogate
        move.w (sp)+, d0
        lea    MenuHandles, a4
        cmp.w  #Aid12ID, d0
        bne    NotAid12
        move.l menuaid12(a4), a3        ;get handle
        bra    AddProgMenu

NotAid12:
        cmp.w  #Aid13ID, d0
        bne    NotAid13
        move.l menuaid13(a4), a3        ;get handle
        bra    AddProgMenu

NotAid13:
        cmp.w  #Aid14ID, d0
        bne    NotAid14
        move.l menuaid14(a4), a3        ;get handle
        bra    AddProgMenu

NotAid14:
        cmp.w  #SS15ID, d0

```

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```

        bne    NotSS15
        move.l menuSS15(a4),a3      ;get handle
        bra    AddProgMenu

NotSS15:
        move.l menunone(a4),a3
        move.w #20,-(sp)
        _SysBeep

AddProgMenu:
        move.w #NoneMenu,-(sp)
        _DeleteMenu
        move.l a3,-(sp)
        move.w #0,-(sp)
        _InsertMenu
;redraw the bar
        _DrawMenuBar

ClearReturn:
        move.w #0,-(sp)            ;clear any highlighting.
        _HiLiteMenu
        rts

```

```

; Name: MenuBar
; Function: This routine should be called when the mouse is clicked in the
;           menu bar.
; Input: None
; Output: None
MenuBar:

```

```

        clr.l  -(sp)                ;space for result
        move.l where,-(sp)          ;location of mouse
        _MenuSelect
        move.l (sp)+,d0             ;get result (menu id, item #)
        swap  d0                    ;get menu id in low word

```

```

Choices:
        cmp.w  #0,d0                ;Was it in any menu?
        beq    @1                    ;no menu id
        cmp.w  #AppleMenu,d0        ;Was it in the apple menu?
        beq    InAppleMenu
        cmp.w  #FileMenu,d0         ;Was it in the file menu?
        beq    InFileMenu
        cmp.w  #NoneMenu,d0
        beq    InSSMenu
        cmp.w  #Aid12Menu,d0
        beq    InAidMenu
        cmp.w  #Aid13Menu,d0
        beq    InAidMenu
        cmp.w  #Aid14Menu,d0
        beq    InAidMenu
        cmp.w  #SS15Menu,d0
        beq    InSSMenu
        @1    bra    ClearReturn

```

```

InAppleMenu:
; GetItem

```

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```

swap    d0                                ; get item # in low word
cmp.w   #AboutItem,d0
bne     NotAbout

; Open About dialog window.
; FUNCTION NewWindow (wStorage: Ptr; boundsRect: Rect;
;                     title: Str255; visible: BOOLEAN;
;                     procID: INTEGER; behind: WindowPtr;
;                     goAwayFlag: BOOLEAN;
;                     refCon: LongInt) : WindowPtr;
SUBQ    #4,SP                                ; Space for function result
CLR.L   -(SP)                                ; Storage for window (Heap)
PEA     AboutBounds                          ; Window position
PEA     'About WDHA'                        ; Window title
MOVE.B  #255,-(SP)                          ; Make window visible
MOVE    #dBoxProc,-(SP)                    ; Standard document window
MOVE.L  #-1,-(SP)                          ; Make it the front window
move.B  #-1,-(SP)                          ; Window has goAway button
CLR.L   -(SP)                                ; Window refCon
_NewWindow
lea     AboutPtr,a4
MOVE.L  (SP)+,(a4)                          ; Save handle for later
MOVE.L  (a4),-(SP)                          ; Make sure the new window is the port
; PROCEDURE SetPort (gp: GrafPort)
_SetPort                                ; Make it the current port
move.w  #0,-(sp)
_TextFont                                ; Make sure it's the system font
move.w  #1,-(sp)                          ; Bold
_TextFace
DispString #20,#16,Wearable Digital Hearing Aid Fitting Procedure V. 1.0
move.w  #0,-(sp)                          ; Plain Text
_TextFace
DispString #200,#32,Central Institute For The Deaf
DispString #200,#48,818 South Euclid Ave.
DispString #200,#64,St. Louis Mo. 63110
DispString #200,#80,Phone: 314-652-3200
move.w  #1,-(sp)                          ; Bold
_TextFace
DispString #20,#96,Supported in part by:
-move.w  #0,-(sp)                          ; Plain Text
_TextFace
DispString #40,#112,The Rehabilitation Research And Development Service
DispString #40,#128,Dept. of Medicine and Surgery: Veterans Administration
; Print the big "CID"
move.w  #36,-(sp)
_TextSize
move.w  #17,-(sp)                          ; Bold+Shadow
_TextFace
DispString #44,#64,CID
; Set text characteristics back to normal
move.w  #12,-(sp)
_TextSize
move.w  #0,-(sp)                          ; Plain Text
_TextFace
; Wait for an event

```

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```

        move.l #0000FFFF,d0
        _FlushEvents
EvtWait:
; FUNCTION      GetNextEvent(eventMask: INTEGER;
;               VAR theEvent: EventRecord) : BOOLEAN
;
        CLR        -(SP)                ; Clear space for result
        MOVE    #$000F,-(SP)            ; Allow 12 low events
        PEA      EventRecord            ; Place to return results
        _GetNextEvent                    ; Look for an event
        MOVE    (SP)+,d0                ; Get result code
        BEQ     EvtWait                  ; No event... Keep waiting
; Dispose Window
        move.l AboutPtr,-(sp)
        _DisposWindow
        bra      ClearReturn
NotAbout:
        lea      MenuHandles,a4
        move.l menuapple(a4),-(sp)      ; Look in Apple Menu
        move.w d0,-(sp)                ; what item #
        pea     DeskName                ; get item name
        _GetItem
; OpenDeskAcc
        clr.w    -(sp)                  ; space for result
        pea     DeskName                ; open DeskName acc
        _OpenDeskAcc
        move.w (sp)+,d0                  ; pop result
        bra      ClearReturn

InFileMenu:
        swap     d0                      ; get item # in low word
        cmp.w    #QuitItem,d0           ; Is it quit?
        bne      DoneFile                ; If not forget it
        bsr      WDHAPSClose             ; dispose of the parameter settings window
        bsr      WDHATCClose             ; dispose of the test/calibrate window
        _ExitToShell                     ; leave application
DoneFile:
        bra      ClearReturn

InAidMenu:
        swap     d0                      ; get item # in low word
        cmp.w    #NewProgItem,d0
        bne      @9
        bsr      SetProgMenu
        bra      WMDone
@9
        cmp.w    #SetItem,d0
        bne      @1
        bsr      WDHAPSShow
        bra      WMDone
@1
        cmp.w    #TestItem,d0
        bne      @2
        bsr      WDHATCShow
        bra      WMDone
@2
        cmp.w    #FCItem,d0

```

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```

        bne    @4
        bsr    WDHAFCSet
        bra    WMDone
@4:
WMDone    bra    ClearReturn

```

```

InSSMenu:
        swap   d0                ; get item # in low word
        cmp.w  #NewProgItem,d0
        bne    @1
        bsr    SetProgMenu
        bra    SSDone
@1:     cmp.w  #LoadItem,d0
        bne    @2
        bsr    WDHASetFileParams
        bra    SSDone
@2:
SSDone  bra    ClearReturn

```

```

;-----Data starts here-----
MenuHandles:
        dc.l   0                ;handle to apple menu
        dc.l   0                ;handle to file menu
        dc.l   0                ;handle to aid12 menu
        dc.l   0                ;handle to aid13 menu
        dc.l   0                ;handle to aid14 menu
        dc.l   0                ;handle to ss15 menu
        dc.l   0                ;handle to none menu

AppleName:  dc.b   1,$14        ; A string containing the apple symbol
DeskName:   dcb.w  16,0         ;desk accessories name

AboutPtr    dc.l   0            ; the About dialog window pointer
AboutBounds:
        dc.w   100             ; upper
        dc.w   50              ; left
        dc.w   232             ; lower
        dc.w   472             ; right

```

;WDHAMenu header file

; This file must be included if any routines in WDHAMenu are used.

xref MakeMenus

xref MenuHandles

xref MenuBar

```
; file WDHAPS.Asm
```

```
Include MacTraps.D
Include ToolEqu.D
Include SysEquX.D
Include QuickEquX.D
Include SANEMacs.txt
Include MDS2:WDHA.hdr
Include MDS2:WDHASCSI.hdr
```

```
-----
: WDHA Paramater Settings Window Manager
```

```
: This package contains routines to manipulate the WDHA Parameter
: Settings window. This window contains an interface which controls the
: gain and limit of each channel of the WDHA by allowing the user to move
: bars on a graph of Frequency versus dB SPL (execute the program for a better
: understanding), this control is referred to as the "PSGraph" in the program
: documentation. Next to this graph is a chart (the "PSChart") containing the
: numeric values of each channel's gain and limit.
```

```
: It also contains control buttons to specify if the WDHA should be in
: Hearing aid mode, if the input attenuation should be off or on, and whether
: the aid should use the probe mike or the field mike. The output attenuation
: is automatically turned on or off by the program, it's control being used
: as an indicator of this status.
```

```
: Wherever the documentation refers to the term "theta", it is referring
: to the height of the lower bar of the bar graph, and wherever the documentation
: uses "phi", it refers to the height of the upper bar.
```

```
-----External Definitions-----
```

```
XDEF WDHAPSOOpen
XDEF WDHAPSClose
XDEF WDHAPSShow
XDEF WDHAPSHide
XDEF WDHAPSDraw
XDEF WDHAPSControl
XDEF WDHAPSIIS
XDEF WDHAPSSetParam
```

```
----- Constant Definitions -----
: CHANNELS EQU 4 ; There are four channels
```

```
; PSG = The Parameter Settings Graph
```

```
PSGHeight EQU 120 ; Graph height in pixels
PSGChanWidth EQU 20 ; each bar is PSGChanWidth pixels wide.
PSGWidth EQU CHANNELS*PSGChanWidth ; Graph width in pixels
PSGInitX EQU 30 ; initial X coord (local) of ul corner of graph
PSGInitY EQU 20 ; initial Y coord (local) of ul corner of graph
```

```
; PSC = The Parameter Settings Chart
```

```
PSCFWidth EQU 46 ; channel, gain and limit field width
PSCFHeight EQU PSGHeight/(CHANNELS+1) ; height of box in chart
PSCWidth EQU 3*PSCFWidth
PSCInitX EQU PSGInitX+PSGWidth ; X coord (local) of ul corner of chart
```

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PSCInitY EQU PSGInitY ; Y coord (local) of ul corner of chart

; PS = The Parameter Settings Window

PSInitX EQU 60 ; initial X coord (global) of upper left corner

PSInitY EQU 80 ; initial Y coord (global) of upper left corner

PSRight EQU PSInitX+PSGWidth+PSCWidth+2*PSGInitX+140

PSTxtSize EQU 12

; PSCtl = The Control Buttons

PSCtlInitX EQU PSGInitX+PSGWidth+PSCWidth+10

PSCtlInitY EQU PSGInitY+5

PSCtlFHeight EQU PSCFHeight

-----Subroutine Declarations-----

; Name: WDHAPSOOpen

; Function: Call this routine to create and display the PS Window.

; Input: None

; Output: None

WDHAPSOOpen:

movem.l d0-d2/a0-a6,-(sp) ; save registers

; Set up document window.

; FUNCTION NewWindow (wStorage: Ptr; boundsRect: Rect;
title: Str255; visible: BOOLEAN;
procID: INTEGER; behind: WindowPtr;
goAwayFlag: BOOLEAN;
refCon: LongInt) : WindowPtr;

SUBQ #4,SP ; Space for function result

CLRL -(SP) ; Storage for window (Heap)

PEA WDHAPSBounds ; Window position

PEA 'WDHA Parameter Settings' ; Window title

MOVEB #255,-(SP) ; Make window visible

MOVE #rDocProc,-(SP) ; Standard document window

MOVEL #-1,-(SP) ; Make it the front window

move.B #-1,-(SP) ; Window has goAway button

CLRL -(SP) ; Window refCon

_NewWindow ; Create and draw window

lea WDHAPSPtr,a4

MOVEL (SP)+,(a4) ; Save handle for later

MOVEL (a4),-(SP) ; Make sure the new window is the port

; PROCEDURE SetPort (gp: GrafPort)

_SetPort ; Make it the current port

; Add the control buttons

bsr PSAddControls

bsr WDHAPSDraw

movem.l (sp)+,d0-d2/a0-a6 ; Restore registers

RTS

; Name: WDHAPSClose

; Function: Call this routine to destroy the PS Window and remove it from
the screen.

; Input: None

; Output: None

WDHAPSClose:

movem.l d0-d7/a0-a6,-(sp) ; save registers

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```

        move.l WDHAPSPtr,-(sp)
        _KillControls
; Dispose Window
        move.l WDHAPSPtr,-(sp)
        _DisposWindow
        movem.l      (sp)+,d0-d7/a0-a6      ; restore registers
        rts

; Name: WDHAPSShow
; Function: This routine makes the PS window visible and frontmost.
; Input: None
; Output: None
WDHAPSShow:
        movem.l      d0-d7/a0-a6,-(sp)      ; save registers
; Bring it to the front
        move.l WDHAPSPtr,-(sp)
        _BringToFront
; Show Window
        move.l WDHAPSPtr,-(sp)
        _ShowWindow
        move.l WDHAPSPtr,-(sp)
        _SelectWindow      ; So select it.
        movem.l      (sp)+,d0-d7/a0-a6      ; restore registers
        rts

; Name: WDHAPSHide
; Function: This routine makes the PS window invisible, removing it from the
; screen (but not destroying it).
; Input: None
; Output: None
WDHAPSHide:
        movem.l      d0-d7/a0-a6,-(sp)      ; save registers
; Hide Window
        move.l WDHAPSPtr,-(sp)
        _HideWindow
        movem.l      (sp)+,d0-d7/a0-a6      ; restore registers
        rts

; Name: WDHAPSDraw
; Function: This routine draws the PS window's contents.
; Input: None
; Output: None
WDHAPSDraw:
        movem.l      d0-d7/a0-a6,-(sp)      ; save registers
        lea    WDHAPSPtr,a4      ; Pointer on stack
        MOVE.L (a4),-(SP)
; PROCEDURE SetPort (gp: GrafPort)
        _SetPort      ; Make it the current port
; First draw the graph
        pea    WDHAPSGraph
        _EraseRect      ; clear it
        pea    WDHAPSGraph
        _FrameRect      ; Frame it
        move.w #patOr,-(sp)

```

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```

_PenMode                                ; change to Or pen mode.

move.w #0,d4                            ; count thru channels
DrawChans:                              ; draw each channel
    cmp.w #CHANNELS,d4                  ; done yet?
    beq    DoneDC
; Draw Theta Bar
    pea    ThetaPat
    _PenPat                                ; set pen pattern to ThetaPat
    move.wd4,-(sp)
    bsr    CalThetaRect                ; Calculate theta rectangle
    pea    TRect
    _PaintRect                          ; Fill with pattern
; Draw Phi Bar
    pea    PhiPat
    _PenPat                                ; set pen pattern to PhiPat
    move.wd4,-(sp)
    bsr    CalPhiRect
    pea    TRect
    _PaintRect                          ; Fill with pattern
    add.w #1,d4
    bra    DrawChans
DoneDC:
    _PenNormal                          ; Reset Pen to original settings
    move.w #PSTxtSize,-(sp)
    _TextSize
    move.w #PSGInitX+0*PSGChanWidth+PSGChanWidth/2,-(sp)
    move.w #PSGInitY+PSGHeight+PSTxtSize,-(sp)
    _MoveTo
    move.w #'1',-(sp)
    _DrawChar
    move.w #PSGInitX+1*PSGChanWidth+PSGChanWidth/2,-(sp)
    move.w #PSGInitY+PSGHeight+PSTxtSize,-(sp)
    _MoveTo
    move.w #'2',-(sp)
    _DrawChar
    move.w #PSGInitX+2*PSGChanWidth+PSGChanWidth/2,-(sp)
    move.w #PSGInitY+PSGHeight+PSTxtSize,-(sp)
    _MoveTo
    move.w #'3',-(sp)
    _DrawChar
    move.w #PSGInitX+3*PSGChanWidth+PSGChanWidth/2,-(sp)
    move.w #PSGInitY+PSGHeight+PSTxtSize,-(sp)
    _MoveTo
    move.w #'4',-(sp)
    _DrawChar
    move.w #PSGInitX+(CHANNELS/2)*PSGChanWidth-25,-(sp)
    move.w #PSGInitY+PSGHeight+2*PSTxtSize,-(sp)
    _MoveTo
    pea    'Channel'
    _DrawString
    move.w #PSGInitX-20,-(sp)
    move.w #PSGInitY+PSGHeight/2-PSTxtSize,-(sp)
    _MoveTo

```

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```

    pea    'dB'
    _DrawString
    move.w #PSGInitX-24,-(sp)
    move.w #PSGInitY+PSGHeight/2,-(sp)
    _MoveTo
    pea    'SPL'
    _DrawString
    move.w #9,-(sp)
    _TextSize
    move.w #PSGInitX-9,-(sp)
    move.w #PSGInitY+PSGHeight,-(sp)
    _MoveTo
    move.w #'0',-(sp)
    _DrawChar
    move.w #PSGInitX-20,-(sp)
    move.w #PSGInitY+9,-(sp)
    _MoveTo
    pea    '120'
    _DrawString
; Now draw the chart.
    _PenNormal
    pea    WDHAPSChart
    _FrameRect
    move.w #PSCInitX,-(sp)
    move.w #PSCInitY+1*PSCFHeight,-(sp)
    _MoveTo
    move.w #PSCInitX+PSCWidth,-(sp)
    move.w #PSCInitY+1*PSCFHeight,-(sp)
    _LineTo
    move.w #PSCInitX,-(sp)
    move.w #PSCInitY+2*PSCFHeight,-(sp)
    _MoveTo
    move.w #PSCInitX+PSCWidth,-(sp)
    move.w #PSCInitY+2*PSCFHeight,-(sp)
    _LineTo
    move.w #PSCInitX,-(sp)
    move.w #PSCInitY+3*PSCFHeight,-(sp)
    _MoveTo
    move.w #PSCInitX+PSCWidth,-(sp)
    move.w #PSCInitY+3*PSCFHeight,-(sp)
    _LineTo
    move.w #PSCInitX,-(sp)
    move.w #PSCInitY+4*PSCFHeight,-(sp)
    _MoveTo
    move.w #PSCInitX+PSCWidth,-(sp)
    move.w #PSCInitY+4*PSCFHeight,-(sp)
    _LineTo
    move.w #PSCInitX+PSCFWidth,-(sp)
    move.w #PSCInitY,-(sp)
    _MoveTo
    move.w #PSCInitX+PSCFWidth,-(sp)
    move.w #PSCInitY+PSGHeight,-(sp)
    _LineTo
    move.w #PSCInitX+2*PSCFWidth,-(sp)

```

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```

move.w #PSCInitY, -(sp)
_MoveTo
move.w #PSCInitX+2*PSCFWidth, -(sp)
move.w #PSCInitY+PSGHeight, -(sp)
_LineTo
move.w #PSCInitX+6, -(sp)
move.w #PSCInitY+PSCFHeight-6, -(sp)
_MoveTo
pea 'Channel'
_DrawString
move.w #PSCInitX+PSCFWidth+11, -(sp)
move.w #PSCInitY+PSCFHeight-6, -(sp)
_MoveTo
pea 'Gain'
_DrawString
move.w #PSCInitX+2*PSCFWidth+10, -(sp)
move.w #PSCInitY+PSCFHeight-6, -(sp)
_MoveTo
pea 'Limit'
_DrawString
move.w #CHANNELS, d4 ; Now draw the chart data with PrintVal
lea Theta3, a0 ; will draw the gains and limits too

DrChartNums:
; Draw channel #
move.w #0, -(sp) ; Column 0
move.w d4, -(sp) ; Row is same as channel
move.w d4, -(sp) ; value is channel
bsr PrintVal
; Draw gain
move.w #1, -(sp) ; now do gain
move.w d4, -(sp) ; Row is same as channel
move.w (a0), -(sp) ; Show the theta value as gain
bsr PrintVal
; Draw limit
move.w #2, -(sp) ; now do limit
move.w d4, -(sp) ; Row is same as channel
move.w 2(a0), -(sp) ; Show the Phi value as limit
bsr PrintVal
lea -4(a0), a0
sub.w #1, d4
bne DrChartNums
; Draw the control buttons.
move.l WDHA PSPtr, -(sp) ; the window ptr
_DrawControls
bsr WDHA PSetParam ; update the WDHA
movem.l (sp)+, d0-d7/a0-a6 ; restore registers
rts

; Name: PSAddControls
; Function: This routine adds the PS window's controls.
; Input: None
; Output: None
PSAddControls:
movem.l d0-d7/a0-a6, -(sp) ; save registers

```

```

; Set up the controls bounding rectangle.
    lea        TRect,a4
    move.w#PSCtlInitY+0*PSCtlFHeight,(a4)    ; store y coord
    move.w#PSCtlInitX,2(a4)                  ; store x coord
    move.w#PSCtlInitY+0*PSCtlFHeight+20,4(a4) ; store y coord
    move.w#PSRight,6(a4)                     ; store x coord
; Push parameters for NewControl
    clr.l      -(sp)                          ; NewControl returns a handle
    move.l WDHAPSPtr,-(sp)                    ; the window ptr
    pea        TRect                          ; the rectangle bounding the control
    pea        'Hearing Aid On'               ; title
    move.b #TRUE,-(sp)                        ; visible
    move.w#0,-(sp)                            ; value
    move.w#0,-(sp)                            ; min
    move.w#1,-(sp)                            ; max
    move.w#1,-(sp)                            ; check box proc id
    move.l #0,-(sp)                           ; refcon not used
; Call NewControl
    _NewControl
    lea        AidControl,a3
    move.l (sp)+,(a3)                          ; store the result
; Set up the controls bounding rectangle.
    lea        TRect,a4
    move.w#PSCtlInitY+1*PSCtlFHeight,(a4)    ; store y coord
    move.w#PSCtlInitX,2(a4)                  ; store x coord
    move.w#PSCtlInitY+1*PSCtlFHeight+20,4(a4) ; store y coord
    move.w#PSRight,6(a4)                     ; store x coord
; Push parameters for NewControl
    clr.l      -(sp)                          ; NewControl returns a handle
    move.l WDHAPSPtr,-(sp)                    ; the window ptr
    pea        TRect                          ; the rectangle bounding the control
    pea        'Input Attenuation'           ; title
    move.b #TRUE,-(sp)                        ; visible
    move.w#0,-(sp)                            ; value
    move.w#0,-(sp)                            ; min
    move.w#1,-(sp)                            ; max
    move.w#1,-(sp)                            ; check box proc id
    move.l #0,-(sp)                           ; refcon not used
; Call NewControl
    _NewControl
    lea        IAControl,a3
    move.l (sp)+,(a3)                          ; store the result
; Set up the controls bounding rectangle.
    lea        TRect,a4
    move.w#PSCtlInitY+2*PSCtlFHeight,(a4)    ; store y coord
    move.w#PSCtlInitX,2(a4)                  ; store x coord
    move.w#PSCtlInitY+2*PSCtlFHeight+20,4(a4) ; store y coord
    move.w#PSRight,6(a4)                     ; store x coord
; Push parameters for NewControl
    clr.l      -(sp)                          ; NewControl returns a handle
    move.l WDHAPSPtr,-(sp)                    ; the window ptr
    pea        TRect                          ; the rectangle bounding the control
    pea        'Output Attenuation'          ; title
    move.b #TRUE,-(sp)                        ; visible

```

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```

        move.w #0,-(sp)           ; value
        move.w #0,-(sp)           ; min
        move.w #1,-(sp)           ; max
        move.w #1,-(sp)           ; check box proc id
        move.l #0,-(sp)           ; refcon not used
; Call NewControl
    _NewControl
    lea        OAControl,a3
    move.l (sp)+,(a3)             ; store the result
; Set up the controls bounding rectangle.
    lea        TRect,a4
    move.w #PSCtlInitY+3*PSCtlFHeight,(a4) ; store y coord
    move.w #PSCtlInitX,2(a4)      ; store x coord
    move.w #PSCtlInitY+3*PSCtlFHeight+20,4(a4) ; store y coord
    move.w #PSRight,6(a4)        ; store x coord
; Push parameters for NewControl
    clr.l      -(sp)              ; NewControl returns a handle
    move.l WDHAPSPtr,-(sp)        ; the window ptr
    pea        TRect              ; the rectangle bounding the control
    pea        'Field Mike'      ; title
    move.b #TRUE,-(sp)           ; visible
    move.w #1,-(sp)              ; make Field mike on as the default
    move.w #0,-(sp)              ; min
    move.w #1,-(sp)              ; max
    move.w #2,-(sp)              ; radio button proc id
    move.l #0,-(sp)              ; refcon not used
; Call NewControl
    _NewControl
    lea        FieldControl,a3
    move.l (sp)+,(a3)             ; store the result
; Set up the controls bounding rectangle.
    lea        TRect,a4
    move.w #PSCtlInitY+4*PSCtlFHeight,(a4) ; store y coord
    move.w #PSCtlInitX,2(a4)      ; store x coord
    move.w #PSCtlInitY+4*PSCtlFHeight+20,4(a4) ; store y coord
    move.w #PSRight,6(a4)        ; store x coord
; Push parameters for NewControl
    clr.l      -(sp)              ; NewControl returns a handle
    move.l WDHAPSPtr,-(sp)        ; the window ptr
    pea        TRect              ; the rectangle bounding the control
    pea        'Probe Mike'      ; title
    move.b #TRUE,-(sp)           ; visible
    move.w #0,-(sp)              ; value
    move.w #0,-(sp)              ; min
    move.w #1,-(sp)              ; max
    move.w #2,-(sp)              ; radio button proc id
    move.l #0,-(sp)              ; refcon not used
; Call NewControl
    _NewControl
    lea        ProbeControl,a3
    move.l (sp)+,(a3)             ; store the result
    movem.l    (sp)+,d0-d7/a0-a6
    rts

```

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; CalThetaRect calculates the rectangle surrounding the control bar for the
 ; given channel.
 ; Input: the channel # (a word) is passed on the stack.
 ; Output: the rect TRect is filled.

CalThetaRect:

```

movem.l    d0-d7/a0-a6,-(sp)
lea    TRect,a4    ; get address of TRect
move.w #PSGInitY+PSGHeight,d4    ; bottom of graph
move.wd4,4(a4)    ; store it in TRect
lea    Theta0,a3    ; Get theta
move.w64(sp),d3    ; Get channel number
asl.w  #2,d3    ; *4
sub.w  (a3,d3.w),d4    ; compute top of bar y coord
move.wd4,(a4)    ; store it in TRect
move.w64(sp),d3    ; Get channel number
mulu   #PSGChanWidth,d3    ; channel # * ChanWidth
add.w  #PSGInitX,d3    ; move over
move.wd3,2(a4)    ; store left side
add.w  #PSGChanWidth,d3    ; add width
move.wd3,6(a4)    ; store right side
pea    TRect
move.w #1,-(sp)
move.w #1,-(sp)
_insetRect    ; make it a tad smaller
sub.w  #1,(a4)    ; not the top level though
movem.l    (sp)+,d0-d7/a0-a6
move.l (sp),2(sp)    ; move return address over param
tst.w  (sp)+    ; get rid of parameter
rts    ; and return

```

; CalPhiRect calculates the rectangle surrounding the control bar for the
 ; given channel.
 ; Input: the channel # (a word) is passed on the stack.
 ; Output: the rect TRect is filled.

CalPhiRect:

```

movem.l    d0-d7/a0-a6,-(sp)
lea    TRect,a4    ; get address of TRect
move.w #PSGInitY,d4    ; top of graph
move.wd4,(a4)    ; store it in TRect
lea    Phi0,a3    ; Get Phi
move.w64(sp),d3    ; Get channel number
asl.w  #2,d3    ; *4
move.w #120,d5
sub.w  (a3,d3.w),d5    ; compute bottom of bar y coord
add.w  d5,d4
move.wd4,4(a4)    ; store it in TRect
move.w64(sp),d3    ; Get channel number
mulu   #PSGChanWidth,d3    ; channel # * ChanWidth
add.w  #PSGInitX,d3    ; move over
move.wd3,2(a4)    ; store left side
add.w  #PSGChanWidth,d3    ; add width
move.wd3,6(a4)    ; store right side
pea    TRect
move.w #1,-(sp)

```

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```

    move.w #1, -(sp)
    _InsetRect                ; make it a tad smaller
    add.w #1, 4(a4)           ; not the bottom though
    movem.l (sp)+, d0-d7/a0-a6
    move.l (sp), 2(sp)        ; move return address over param
    tst.w (sp)+               ; get rid of parameter
    rts                       ; and return

; Name: PrintVal
; Function: This routine prints the given value at the specified row and
; column of the PSChart.
; Input: d3 (word) = value, d4 = row, d5 = column
; Output: None
PrintVal:
    movem.l d0-d7/a0-a6, -(sp) ; save registers
    move.w64(sp), d3           ; d3 = value to be printed
    move.w66(sp), d4           ; d4 = Row in chart
    move.w68(sp), d5           ; d5 = column in chart
; compute x coord
    mulu #PSCFWidth, d5        ; column * width of each field
    add.w #PSCInitX+24, d5     ; shift over
; compute y coord
    add.w #1, d4               ; add 1 to row
    mulu #PSCFHeight, d4       ; * height of each field
    add.w #PSCInitY-6, d4      ; shift down and then up a little
; erase whatever is there already.
    lea TRect, a2              ; we'll put it in TRect
    move.wd5, 2(a2)             ; our x is the left x
    move.wd5, 6(a2)             ; then compute the right
    add.w #20, 6(a2)            ; as 20 over from the left
    move.wd4, 4(a2)             ; our y is the bottom y
    move.wd4, (a2)              ; then compute the top
    sub.w #PSTxtSize, (a2)      ; as TxtSize up from bottom
    pea TRect                  ; now erase it
    _EraseRect
; move there
    move.wd5, -(sp)
    move.wd4, -(sp)
    _MoveTo
; convert value to string
    move.wd3, d0                ; NumToString expects val in d0
    lea NumBuf, a0              ; address of NumBuf in a0
    move.w #0, -(SP)            ; Select NumToString
    _Pack7
    pea NumBuf
    _DrawString
    movem.l (sp)+, d0-d7/a0-a6
    move.l (sp), 6(sp)          ; move return address over parameters
    add.l #6, sp                ; get rid of parameters
    rts

; Name: WDHAPSIS
; Function: This routine returns a Boolean telling whether or not
; the given window pointer is the PS window's pointer.

```


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; Input: A window pointer (passed on the stack)
 ; Output: a word, TRUE or FALSE (defined in WDHA.hdr) returned on the stack.
 ; **Note: You do not have to push a word for the result of this routine.

WDHAPSPtr:

```

    movem.l      a4/d4,-(sp)      ; save registers
    move.l       8(sp),a4         ; get return address in a4
    move.l       12(sp),d4        ; get WindowPtr in d4
    cmp.l        WDHAPSPtr,d4     ; Was it our window?
    beq          IS10             ; It is
    move.w       #FALSE,14(sp)    ; save result
    bra          IS20
IS10: move.w     #TRUE,14(sp)
IS20: move.l     a4,10(sp)        ; put return address back
    movem.l      (sp)+,a4/d4      ; restore registers
    tst.w        (sp)+           ; get rid of extra two bytes
    rts                                ; return

```

; Name: WDHAPSPtrControl

; Function: This routine should be called whenever a mousedown event occurs

; within the contents of the PS Window. It handles the highlighting of the

; proper control buttons, and sends the proper records to the WDHA.

; Input: The mouse location (on the stack), from the event's where field.

; Output: None

WDHAPSPtrControl:

```

    movem.l      d0-d7/a0-a6,-(sp)
    move.l       WDHAPSPtr,-(sp) ; WDHAPSPtr on stack
;PROCEDURE SetPort (gp: GrafPort)
    _SetPort      ; Make sure it's the current
port

```

```

    pea 64(sp) ; push address of point
    _GlobalToLocal ; convert it to the window's coords
; Was it in a control button?
ButtonCheck:
; call FindControl
    clr.w  -(sp) ; returns a long
    move.l 66(sp),-(sp) ; push point in local coords
    move.l WDHAPSPtr,-(sp) ; WDHAPSPtr on stack
    pea WhichControl ; which one?
    _FindControl
    tst.w (sp)+ ; pop result
    lea WhichControl,a4
    tst.l (a4) ; Was it in any of them?
    beq ChanCheck ; if not try the graph
; if it was in a control, call TrackControl
    clr.w  -(sp) ; returns a word
    move.l WhichControl,-(sp) ; WhichControl now has the handle
    move.l 70(sp),-(sp) ; starting point
    move.l #0,-(sp) ; no action proc
    _TrackControl
    tst.w (sp)+ ; did they change the button?
    beq NoChan ; if not then leave
; Was it the output Attenuation button?
    lea WhichControl,a4

```

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```

        move.l OAControl,d4
        cmp.l (a4),d4
        bne      NotOA      ; if not then was it the IA button?

; It was the output attenuation button so adjust the bar heights.
        clr.w d3      ; use d3 as a channel counter
        lea      Theta0,a3
CGLoop11:
        cmp.w #CHANNELS,d3
        beq      InvBut
        clr.w -(sp)
        bsr      GOUT
        move.w(a3),d0      ; get Theta in d0
        sub.w (sp),d0      ; subtract the old GOUT from Theta
        move.wd0,(a3)      ; store Theta
        move.w2(a3),d1      ; get phi in d1
        sub.w (sp)+,d1      ; subtract the old GOUT from Phi
        move.wd1,2(a3)      ; store phi
        lea      4(a3),a3
        add.w #1,d3
        bra      CGLoop11

InvBut:
        clr.w -(sp)      ; GetCtlValue returns a word
        move.l OAControl,-(sp)
        _GetCtlValue
        move.w(sp)+,d3      ; now value is in d3
        not.w d3
        and.w #1,d3      ; invert the status.
        move.l WhichControl,-(sp)
        move.wd3,-(sp)      ; set it to the new value.
        _SetCtlValue

        clr.w d3      ; use d3 as a channel counter
        lea      Theta0,a3
CGLoop12:
        cmp.w #CHANNELS,d3
        beq      UDScreen
        clr.w -(sp)
        bsr      GOUT
        move.w(a3),d0      ; get Theta in d0
        add.w (sp),d0      ; add the new GOUT
        move.wd3,-(sp)      ; now clip the gain as necessary
        move.wd0,-(sp)      ; the new gain
        bsr      ValidGain
        move.w(sp)+,(a3)      ; store it
        move.w2(a3),d1      ; get phi in d1
        add.w (sp)+,d1      ; add the new GOUT to Phi
        move.wd3,-(sp)      ; now clip the limit as necessary
        move.wd1,-(sp)      ; the new limit
        bsr      ValidLimit
        move.w(sp)+,2(a3)      ; store phi
        lea      4(a3),a3
        add.w #1,d3

```

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```

        bra            CGLoop12
NotOA:
        move.l  IAControl,d4
        lea     WhichControl,a4
        cmp.l   (a4),d4
        bne     OtherBut                ; if not then forget it.
; It was the input attenuation button so adjust the bar heights.
        clr.w   d3                      ; use d3 as a channel counter
        lea     Theta0,a3
CGLoop21:
        cmp.w   #CHANNELS,d3
        beq     InvBut2
        clr.w   -(sp)
        bsr     GIN
; the gain (the limit is not affected)
        move.w  (a3),d0                  ; get theta
        sub.w   (sp)+,d0                 ; subtract the old GIN
        move.wd0,(a3)                   ; store it back
; go to the next channel
        lea     4(a3),a3
        add.w   #1,d3
        bra     CGLoop21
InvBut2:
        clr.w   -(sp)
        move.l  IAControl,-(sp)
        _GetCtlValue
        move.w  (sp)+,d3                ; now value is in d3
        not.w   d3
        and.w   #1,d3                  ; invert the status.
        move.l  WhichControl,-(sp)
        move.wd3,-(sp)                 ; set it to the new value.
        _SetCtlValue
        clr.w   d3                      ; use d3 as a channel counter
        lea     Theta0,a3
CGLoop22:
        cmp.w   #CHANNELS,d3
        beq     UDScreen
        clr.w   -(sp)
        bsr     GIN
        move.w  (a3),d0                  ; get theta
        add.w   (sp)+,d0                 ; add the new GIN
        move.wd3,-(sp)                 ; now clip the gain as necessary
        move.wd0,-(sp)                 ; the new gain
        bsr     ValidGain
        move.w  (sp)+,(a3)              ; store it
; go to the next channel
        lea     4(a3),a3
        add.w   #1,d3
        bra     CGLoop22
UDScreen
        bsr     WDHAPSDraw

```

```

        bra            NoChan

; invert the control value
OtherBut:
        clr.w    -(sp)
        move.l    WhichControl, -(sp)
        _GetCtlValue
        move.w    (sp)+, d3
        not.w     d3
        and.w     #1, d3
        move.l    WhichControl, -(sp)
        move.wd3, -(sp)
        _SetCtlValue
; Was it the Field button?
        move.l    FieldControl, d4
        lea        WhichControl, a4
        cmp.l     (a4), d4
        bne                    ; if not then forget it
; Otherwise invert off the Probe mike
        clr.w    -(sp)
        move.l    ProbeControl, -(sp)
        _GetCtlValue
        move.w    (sp)+, d3
        not.w     d3
        and.w     #1, d3
        move.l    ProbeControl, -(sp)
        move.wd3, -(sp)
        _SetCtlValue
        bra        NoChan
; Was it the Probe button?
NotField:
        move.l    ProbeControl, d4
        lea        WhichControl, a4
        cmp.l     (a4), d4
        bne                    ; if not then forget it
; Otherwise invert the Field mike
        clr.w    -(sp)
        move.l    FieldControl, -(sp)
        _GetCtlValue
        move.w    (sp)+, d3
        not.w     d3
        and.w     #1, d3
        move.l    FieldControl, -(sp)
        move.wd3, -(sp)
        _SetCtlValue
        bra        NoChan

ChanCheck:
        move.w    #0, d4
        lea        Theta0, a4
FindChan:
        cmp.w     #CHANNELS, d4
        beq        NoChan
; Is it a theta bar?
; count thru channels
; draw each channel
; done yet?

```

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```

    move.wd4,-(sp)
    bsr    CalThetaRect      ; Calculate theta rectangle
    clr.w  -(sp)             ; make room for result
    move.l 66(sp),-(sp)      ; push mouse point
    pea    TRect             ; theta rect in TRect
    _PtInRect
    tst.w  (sp)+
    bne    FoundTheta

; Is it a phi bar?
    lea    2(a4),a4
    move.wd4,-(sp)
    bsr    CalPhiRect        ; Calculate theta rectangle
    clr.w  -(sp)             ; make room for result
    move.l 66(sp),-(sp)      ; push mouse point
    pea    TRect
    _PtInRect
    tst.w  (sp)+
    bne    FoundPhi
    lea    2(a4),a4
    add.w  #1,d4
    bra    FindChan

; a4 points to Theta, d4 contains the channel number.
FoundTheta:
    pea    ThetaPat
    _PenPat
    move.w(a4),d3            ; hold onto original theta
; While the button is down move the bar around, changing theta
FTLoop:
    clr.w  -(sp)             ; Make room for result
    _StillDown               ; Is the button still down?
    tst.w  (sp)+
    beq    NoChan            ; If not then exit otherwise...
; Get the point
    pea    TPoint
    _GetMouse                ; Get mouse location
; First Erase Old Bar
    move.w#patBic,-(sp)
    _PenMode
    move.wd4,-(sp)
    bsr    CalThetaRect
    pea    TRect
    _PaintRect
; Now change the theta parameter
    move.w64(sp),d5           ; the vertical coordinate of start point
    sub.w  TPoint,d5          ; original y - current y
; this will be a negative value if they move down
    move.wd3,(a4)             ; restore original theta
    add.w  d5,(a4)            ; change theta
; Is it OK?
    move.wd4,-(sp)            ; channel #
    move.w(a4),-(sp)          ; gain
    bsr    ValidGain          ; make sure gain is in range
    move.w(sp)+,(a4)

```

; Now draw the new bar

ThDrBar:

```
    move.w #patOr, -(sp)
    _PenMode
    move.w d4, -(sp)
    bsr    CalThetaRect
    pea    TRect
    _PaintRect
```

; Now update the chart value.

```
    cmp.w (a4), d3 ; is there any difference?
    beq    FTLoop    ; If not then don't bother
    move.w #1, -(sp) ; gain column in chart
    move.w d4, -(sp) ; row is channel #
    add.w #1, (sp); + 1
    move.w (a4), -(sp) ; value
    bsr    PrintVal
    bra    FTLoop
```

; a4 points to Phi, d4 contains the channel number.

FoundPhi:

```
    pea    PhiPat
    _PenPat
    move.w (a4), d3 ; store old Phi
```

; While the button is down move the bar around, changing theta.

FPLoop:

```
    clr.w -(sp) ; Make room for result
    _StillDown ; Is the button still down?
    tst.w (sp)+
    beq    NoChan ; If not then exit otherwise...
```

; Get the point

```
    pea    TPoint
    _GetMouse ; Get mouse location
```

; First Erase Old Bar

```
    move.w #patBic, -(sp)
    _PenMode
    move.w d4, -(sp)
    bsr    CalPhiRect
    pea    TRect
    _PaintRect
```

; Now change the Phi parameter

```
    move.w 64(sp), d5 ; the vertical coordinate of start point
    sub.w TPoint, d5 ; original y - current y
```

; this will be a negative value if they move down

```
    move.w d3, (a4) ; restore original Phi
    add.w d5, (a4) ; change Phi
```

; Is it OK?

```
    move.w d4, -(sp) ; channel #
    move.w (a4), -(sp) ; limit
    bsr    ValidLimit ; make sure limit in range
    move.w (sp)+, (a4)
```

; Now draw the new bar

PhiDrBar:

; Now draw the new bar

```
    move.w #patOr, -(sp)
```

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```

_PenMode
move.w d4, -(sp)
bsr    CalPhiRect
pea    TRect
_PaintRect
; Now update the chart value.
cmp.w  (a4), d3 ; is there any difference?
beq    FPLoop    ; If not then don't bother
move.w #2, -(sp) ; limit column in chart
move.w d4, -(sp) ; row is channel #
add.w  #1, (sp); + 1
move.w (a4), -(sp) ; value
bsr    PrintVal
bra    FPLoop

NoChan:
_PenNormal
bsr    WDHAPSSetParam    ; update any changes made to the WDHA.
movem.l    (sp)+, d0-d7/a0-a6
move.l (sp)+, (sp)    ; get rid of param
rts

; Name: WDHAPSSetParam
; Function: This routine sets the WDHA to the parameters set in the WDHA
; window.
; Input: None
; Output: None
WDHAPSSetParam:
    movem.l    d0-d7/a0-a6, -(sp)    ; save registers
; Fill all fields of the paramrec except the gain/input select word.
    bsr    CalcGainsLimits; calculate the gains and limits.
; Now calculate the select word by looking at the control buttons.
    lea    paramrec, a4    ; get the gain/input select word
    move.w #16(a4), d4    ; get the gain input select word
    SPIA:    ; set input attenuation bit
        clr.w  -(sp)    ; GetCtlValue returns a word
        move.l IAControl, -(sp); the handle
        _GetCtlValue
        tst.w  (sp)+
        beq    SPNoIA
    SPDofIA:
        bset.l #INPUT, d4
        bra    SPOA
    SPNoIA:
        bclr.l #INPUT, d4
    SPOA:    ; set output attenuation bit
        clr.w  -(sp)    ; GetCtlValue returns a word
        move.l OAControl, -(sp); the handle
        _GetCtlValue
        tst.w  (sp)+
        beq    SPNoOA
    SPDofOA:
        bset.l #OUTPUT, d4
        bra    SPField
    SPNoOA:

```

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```

        bclr.l  #OUTPUT,d4
SPField:
        clr.w   -(sp)
        move.l  FieldControl,-(sp)
        _GetCtlValue
        tst.w   (sp)+
        beq     SPNoField
SPDoField:
        bset.l  #FIELD,d4
        bra     SPProbe
SPNoField:
        bclr.l  #FIELD,d4
SPProbe:
        clr.w   -(sp)
        move.l  ProbeControl,-(sp)
        _GetCtlValue
        tst.w   (sp)+
        beq     SPNoProbe
SPDoProbe:
        bset.l  #PROBE,d4
        bra     SPSendParams
SPNoProbe:
        bclr.l  #PROBE,d4
SPSendParams:
        move.wd4,16(a4)
; Now send the parameters to the WDHA
        lea     paramrec,a0
        bsr     SetParam
; now wait a little while the WDHA does it's thing.
        move.l  #10000,d1
SPWait:
        sub.l   #1,d1
        bne     SPWait
; Now put the WDHA in either hearing aid state or idle state depending on
; the status of the "Hearing Aid On" button.
        clr.w   -(sp)
        move.l  AidControl,-(sp)
        _GetCtlValue
        tst.w   (sp)+
        beq     SPAidOff
        move.w #-1,d0
        bra     SPSetMode
SPAidOff:
        move.w #-100,d0
SPSetMode:
        jsr     scsiwr
SPDone:
        movem.l (sp)+,d0-d7/a0-a6
        rts

```

; Name: CalcGainsLimits

; Function: Compute the gains and limits fields of the paramrec from


```

; the heights of the theta and phi bars of the bar graph, and the status of
; the attenuation control buttons.
; Input: None
; Output: None
;
; If any of the gains or limits produce an out of range value the
; variable called 'Clipped' will have a non-zero value upon return.
;

```

```

movem.l      a0-a6/d0-d7,-(sp)
lea          Clipped,a1
clr.w       (a1)
lea          Theta0,a4                                ; theta0 here
lea          paramrec,a2                              ; gain0 here
lea          H0,a3
move.w      #CHANNELS,d6                              ; loop through four channels

```

```

        move.w(a4),d4                ; get theta0 (= So)
        sub.w  (a3),d4               ; subtract He
        sub.w  8(a3),d4              ; subtract Hr
        sub.w  #60,d4                ; subtract GIN
        clr.w  -(sp)                 ; subtract GIN
        bsr    GIN
        sub.w  (sp)+,d4
        clr.w  -(sp)                 ; subtract GOUT
        bsr    GOUT
        sub.w  (sp)+,d4

```

```

DoLimit:
    move.w 2(a4),d5           ; Get height (=So lim) in d5
    sub.w  d4,d5              ; Subtract Gd
    sub.w  8(a3),d5           ; subtract Hr
    clr.w  -(sp)              ; subtract GOUT
    bsr                                GOUT
    sub.w  (sp)+,d5

```

```

; but first store Gd and Ld
    move.w    d4,(a6)          ; store Gd
    move.w    d5,2(a6)        ; store Ld
    lea       arg1,a0
    move.w    d4,(a0)          ; store gain (dB) in arg1
    pea       arg1             ;dB gain
    pea       arg4             ;fpdB gain
    FI2X                      ;convert from integer to extended fp
    pea       fp20dB_e         ;20 * log base 10 of e = 8.685889638
    pea       arg4             ;fpdB gain
    fdivx     ;db/fp20dbe (result in arg4)
    pea       arg4
    fexp      ;base e exponential (db ratio in arg4)
    pea       twoex14          ;scale it *2E16 to convert it to fixed point
    pea       arg4
    fmulx
    pea       arg4
    pea       arg1

```

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```

fx2i                ;convert extended to integer
move.w arg1,(a2)    ; store the gain
move.w arg1,d1      ; get the gain
cmp.w #16384,d1
bls                 DCDoLimit
move.w #16384,(a2)  ; store the gain
lea         Clipped,a1
add.w #1,(a1).

; Now the limit
DCDoLimit:
lea         arg1,a0
move.w      d5,(a0) ; store limit (dB) in arg1
pea arg1      ;dB limit
pea arg4      ;fpdB limit
F12X        ;convert from integer to extended fp
pea fp20dB    ;20 * log base 10 of e = 8.685889638
pea arg4      ;fpdB limit
fdivx       ;db/fp20dbe (result in arg4)
pea arg4
fexp        ;base e exponential (db ratio in arg4)
pea arg4
pea arg1
pea twoex14   ;scale it *2E16 to convert it to fixed point
pea arg4
fmulx
fx2i                ;convert extended to integer
move.w arg1,2(a2)   ; store the limit
bpl         DCFinLoop
move.w #32767,2(a2)

; Store them in the paramrec
DCFinLoop:
lea         4(a4),a4 ; go to next theta/phi pair.
lea         4(a2),a2 ; go to next gain/limit pair
lea         2(a3),a3 ; go to next He and Hr
subq.b #1,d6
bne         DCLoop
movem.l     (sp)+,a0-a6/d0-d7
rts

; Name: GIN
; Function: This routine returns the input gain as determined by the
; input attenuation control button, either +0 (on), or +18 (off).
; Input: None
; Output: A word on the stack is filled with the result (the user pushes this)
GIN: movem.l a0-a6/d0-d7,-(sp)
; if input attenuation is on then return 0 otherwise 18
clr.w -(sp) ; make room for result
move.l IAControl,-(sp)
_GetCtlValue
tst.w (sp)+
bne GINon
move.w #18,64(sp)
bra GINDone

GINon

```

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```

        move.w #0,64(sp)
GinDone
        movem.l      (sp)+,a0-a6/d0-d7
        rts

; Name: GOUT
; Function: This routine returns the output gain as determined by the
; output attenuation control button, either -34 (on), or -9 (off).
; Input: None
; Output: A word on the stack is filled with the result (the user pushes this)
GOUT:  movem.l      a0-a6/d0-d7,-(sp)
; if output gain is on then return -34 otherwise -9
        clr.w  -(sp)          ; make room for result
        move.l OACControl,-(sp)
        _GetCtlValue
        tst.w  (sp)+
        bne    GoutOn
        move.w #-9,64(sp)
        bra    GoutDone
GoutOn
        move.w #-34,64(sp)
GoutDone
        movem.l      (sp)+,a0-a6/d0-d7
        rts

```

```

; Name: GMAX
; Function: This routine returns the maximum gain for the given channel.
; Input: The channel number is passed on the stack as a word (0-3).
; Output: The result is on the stack upon return.
; ***Note: You do not have to make room for the result on the stack.
GMAX:

```

```

        movem.l      a0-a6/d0-d7,-(sp)
        move.w #60,d0 ; hold result in d0
        clr.w  -(sp)
        bsr    GIN
        add.w  (sp)+,d0 ; add GIN
        clr.w  -(sp)
        bsr    GOUT
        add.w  (sp)+,d0 ; add GOUT
        lea    He,a0
        move.w 64(sp),d1 ; get channel #
        asl.w  #1,d1 ; *2 for words
        add.w  (a0,d1.w),d0 ; add He
        add.w  8(a0,d1.w),d0 ; add Hr
        move.w d0,64(sp) ; write the result over the parameter
        movem.l      (sp)+,a0-a6/d0-d7
        rts

```

```

; Name: ValidGain
; Function: This routine clips the given gain (bar height) as needed for the
; given channel.
; Input: The channel number and gain passed on the stack as words.
; Output: The result is on top of the stack upon return.
; ***Note: You do not have to make room for the result on the stack.

```

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```

ValidGain:
    movem.l    a0-a6/d0-d7,-(sp)
    move.w66(sp),d0        ; get the channel #
    move.w64(sp),d1        ; get the unclipped gain
    cmp.w #2,d1            ; IS it bigger than the minimum height?
    bge        GainOK1
    move.w#2,d1            ; make it bigger
    bra        VGDone

GainOK1:
    move.wd0,-(sp)          ; get GMAX
    bsr        GMAX
    cmp.w (sp)+,d1
    ble        VGDone
    move.w-2(sp),d1          ; make it GMAX

VGDone:
    move.wd1,66(sp)
    movem.l    (sp)+,a0-a6/d0-d7
    move.l (sp),2(sp)        ; move return address
    tst.w (sp)+            ; get rid of extra word
    rts

```

; Name: LMAX
 ; Function: This routine returns the maximum limit for the given channel.
 ; Input: The channel number is passed on the stack as a word (0-3).
 ; Output: The result is on the stack upon return.
 ; ***Note: You do not have to make room for the result on the stack.

```

LMAX:
    movem.l    a0-a6/d0-d7,-(sp)
    clr.w -(sp)
    bsr        GOUT
    move.w(sp)+,d0          ; add GOUT
    lea        Hr,a0
    move.w64(sp),d1        ; get channel #
    asl.w #1,d1            ; *2 for words
    add.w (a0,d1.w),d0      ; add Hr
    move.wd0,64(sp)        ; write the result over the parameter
    movem.l    (sp)+,a0-a6/d0-d7
    rts

```

; Name: ValidLimit
 ; Function: This routine clips the given limit (bar height) as needed for the
 ; given channel.
 ; Input: The channel number and gain passed on the stack as words.
 ; Output: The result is on top of the stack upon return.
 ; ***Note: You do not have to make room for the result on the stack.

```

ValidLimit:
    movem.l    a0-a6/d0-d7,-(sp)
    move.w66(sp),d0        ; get the channel #
    move.w64(sp),d1        ; get the unclipped limit
    cmp.w #2,d1            ; IS it bigger than the minimum height?
    bge        LimitOK1
    move.w#2,d1            ; make it bigger
    bra        VLDone

LimitOK1:

```

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```

        move.wd0,-(sp)          ; get LMAX
        bsr      LMAX
        cmp.w (sp)+,d1
        ble      VLDone
        move.w-2(sp),d1         ; make it LMAX
VLDone:
        move.wd1,66(sp)
        movem.l  (sp)+,a0-a6/d0-d7
        move.l (sp),2(sp)       ; move return address
        tst.w (sp)+            ; get rid of extra word
        rts

```

```

; -----WDHAPS data declarations-----

```

```

.align 4      ; align to long word boundary
WDHAPSPtr: DC.L 0      ; WDHAPS WindowPtr
AidControl:  DC.L 0      ; Hearing Aid On Control
IAControl:   DC.L 0      ; Input Attenuation Control
OAControl:   DC.L 0      ; Output Attenuation
FieldControl: DC.L 0      ; Field Mike Control
ProbeControl: DC.L 0      ; Probe Mike Control

```

```

.align 2      ; align to word boundary
Theta0: DC.W 50
Phi0:   DC.W 70
Theta1: DC.W 50
Phi1:   DC.W 70
Theta2: DC.W 50
Phi2:   DC.W 70
Theta3: DC.W 50
Phi3:   DC.W 70

```

```

paramrec:      ;WDHA parameter record
               dc.w 16384 ;channel 0 gain
               dc.w 32767 ;channel 0 limit
               dc.w 16384 ;channel 1 gain
               dc.w 32767 ;channel 1 limit
               dc.w 16384 ;channel 2 gain
               dc.w 32767 ;channel 2 limit
               dc.w 16384 ;channel 3 gain
               dc.w 32767 ;channel 3 limit
               dc.w 4224  ;gain/input select word

```

```

He:
      dc.w -100 ;channel 0
      dc.w -95  ;channel 1
      dc.w -90  ;channel 2
      dc.w -84  ;channel 3

```

```

; The He table must(!) follow the He table.

```

```

Hr:
      dc.w 121  ;channel 0
      dc.w 117  ;channel 1
      dc.w 127  ;channel 2

```

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```

dc.w 120 ;channel 3

WDHAPSBounds: ; Bounding rect for window
DC.W PSInitY
DC.W PSInitX
DC.W PSInitY+PSGHeight+PSGInitY+2*PSTxtSize+4
DC.W PSRight

WDHAPSGraph: ; bounding rectangle for graph
DC.W PSGInitY
DC.W PSGInitX
DC.W PSGInitY+PSGHeight
DC.W PSGInitX+PSGWidth

WDHAPSChart: ; bounding rectangle for chart
DC.W PSCInitY
DC.W PSCInitX
DC.W PSCInitY+PSGHeight
DC.W PSCInitX+PSCWidth

TRect:
DC.L 0
DC.L 0 ;For calculating various rectangles.

TPoint: DC.L 0 ;For calculating mouse change.

WhichControl: DC.L 0 ; A control handle, for temporary storage.

ThetaPat: DC.B $AA,$55,$AA,$55,$AA,$55,$AA,$55
PhiPat: DC.B $55,$AA,$55,$AA,$55,$AA,$55,$AA

NumBuf: DCB.B 64,0 ; Buffer for number conversion

arg1 dcb.w 8,0 ;integer buffer
arg2 dcb.w 8,0 ;extended floating point buffer
arg3 dcb.w 8,0 ;extended floating point buffer
arg4 dcb.w 8,0 ;extended floating point buffer
arg5 dcb.w 8,0 ;extended floating point buffer
twoex14 dc.w $400d,$8000,$0000,$0000,$0000
fp20dBc dc.w $4002,$8af9,$db22,$d0e5,$6042

Clipped dc.w 0

```

; WDHAPS.hdr
; This file must be included if your program uses the
; WDMA Parameter Settings window.
XREF WDHAPSOOpen
XREF WDHAPSClose
XREF WDHAPSShow
XREF WDHAPSHide
XREF WDHAPSDraw
XREF WDHAPSControl
XREF WDHAPSIIS
XREF WDHAPSSetParam

```
; file WDHATC.Asm
```

```
Include MacTraps.D
Include ToolEqu.D
Include SysEquX.D
Include QuickEquX.D
Include SANEMacs.txt
Include MDS2:WDHA.hdr
Include MDS2:WDHAMac.txt
Include MDS2:WDHASCSI.hdr
```

```
-----
; WDHA Test/Calibrate Window Manager
;   This package contains routines to manipulate the WDHA Test/Calibrate
;   window, which allows you to do pure tone audiometry via the WDHA.
;   The window contains text boxes which allow the user to change the
;   parameters to the test procedure, as well as the control boxes (as in the
;   parameter settings window) to determine the gain/select input word and
;   the on/off status of the hearing aid.
```

```
-----External Definitions-----
```

```
XDEF WDHATCOpen
XDEF WDHATCClose
XDEF WDHATCShow
XDEF WDHATCHide
XDEF WDHATCDraw
XDEF WDHATCControl
XDEF WDHATCIdle
XDEF WDHATCKey
XDEF WDHATCIS
XDEF WDHATCDoTest
```

```
----- Constant Definitions -----
```

```
; TC = The Test/Calibrate Window
TCInitX EQU 30 ; initial X coord (global) of upper left corner
TCInitY EQU 50 ; initial Y coord (global) of upper left corner
TCRightEQU 448
TCTxtSize EQU 12
```

```
; TCctl = The Control Buttons
TCctlInitX EQU 258
TCctlInitY EQU 15
TCctlFHeight EQU 24
```

```
; Text Edit Box Constants
ToneBursts EQU 0
RiseCount EQU 1
OnCount EQU 2
FallCount EQU 3
OffCount EQU 4
Frequency EQU 5
Attenuate EQU 6
```


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TextBoxes EQU 7 ; There are seven boxes

-----Subroutine Declarations-----

; Name: WDHATCOpen

; Function: Call this routine to create and display the TC Window.

; Input: None

; Output: None

WDHATCOpen:

movem.l d0-d2/a0-a6,-(sp) ; save registers

; Set up document window.

; FUNCTION NewWindow (wStorage: Ptr; boundsRect: Rect;
; title: Str255; visible: BOOLEAN;
; procID: INTEGER; behind: WindowPtr;
; goAwayFlag: BOOLEAN;
; refCon: LongInt) : WindowPtr;

SUBQ #4,SP ; Space for function result
CLRL -(SP) ; Storage for window (Heap)
PEA WDHATCBounds ; Window position
PEA 'WDHA Test/Calibrate' ; Window title
MOVEB #255,-(SP) ; Make window visible
MOVE #rDocProc,-(SP) ; Standard document window
MOVE #-1,-(SP) ; Make it the front window
move.b #-1,-(SP) ; Window has goAway button
CLRL -(SP) ; Window refCon
_NewWindow ; Create and draw window

lea WDHATCPtr,a4

MOVEL (SP)+,(a4) ; Save handle for later

MOVEL (a4),-(SP) ; Make sure the new window is the port

; PROCEDURE SetPort (gp: GrafPort)

_SetPort ; Make it the current port

; Add the text boxes.

bsr TCAddBoxes

; Add the control buttons.

bsr TCAddControls

; Draw the content region

bsr WDHATCDraw

movem.l (sp)+,d0-d2/a0-a6 ; Restore registers

RTS

; Name: WDHATCClose

; Function: Call this routine to destroy the TC Window and remove it from
; the screen.

; Input: None

; Output: None

WDHATCClose:

movem.l d0-d7/a0-a6,-(sp) ; save registers

move.l WDHATCPtr,-(sp)

_KillControls

; Dispose Window

move.l WDHATCPtr,-(sp)

_DisposWindow

movem.l (sp)+,d0-d7/a0-a6 ; restore registers

rts

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```

; Name: WDHATCShow
; Function: This routine makes the TC window visible and frontmost.
; Input: None
; Output: None
WDHATCShow:
    movem.l    d0-d7/a0-a6,-(sp)    ; save registers
; Bring it to the front
    move.l    WDHATCPtr,-(sp)
    _BringToFront
; Show Window
    move.l    WDHATCPtr,-(sp)
    _ShowWindow
    move.l    WDHATCPtr,-(sp)
    _SelectWindow
    movem.l    (sp)+,d0-d7/a0-a6    ; restore registers
    rts

; Name: WDHATCHide
; Function: This routine makes the TC window invisible, removing it from the
; screen (but not destroying it).
; Input: None
; Output: None
WDHATCHide:
    movem.l    d0-d7/a0-a6,-(sp)    ; save registers
; Hide Window
    move.l    WDHATCPtr,-(sp)
    _HideWindow
    movem.l    (sp)+,d0-d7/a0-a6    ; restore registers
    rts

; Name: WDHATCDraw
; Function: This routine draws the TC window's contents.
; Input: None
; Output: None
WDHATCDraw:
    movem.l    d0-d7/a0-a6,-(sp)    ; save registers
    lea    WDHATCPtr,a4            ; Pointer on stack
    MOVE.L    (a4),-(SP)
; PROCEDURE SetPort (gp: GrafPort)
    _SetPort                        ; Make it the current port
; Draw the text buttons.
    bsr    TCDrawBoxes
; Draw the control buttons.
    move.l    WDHATCPtr,-(sp)        ; the window ptr
    _DrawControls
    movem.l    (sp)+,d0-d7/a0-a6    ; restore registers
    rts

; Name: TCAddControls
; Function: This routine adds the TC window's controls.
; Input: None
; Output: None
TCAddControls:
    movem.l    d0-d7/a0-a6,-(sp)    ; save registers

```

```

; Set up the controls bounding rectangle.
    lea        TRect,a4
    move.w#TCctlInitY+0*TCctlFHeight,(a4)    ; store y coord
    move.w#TCctlInitX,2(a4)                  ; store x coord
    move.w#TCctlInitY+0*TCctlFHeight+20,4(a4) ; store y coord
    move.w#TCRight,6(a4)                      ; store x coord
; Push parameters for NewControl
    clr.l      -(sp)                          ; NewControl returns a handle
    move.l WDHATCPtr,-(sp)                    ; the window ptr
    pea        TRect                          ; the rectangle bounding the control
    pea        'Hearing Aid On'               ; title
    move.b #TRUE,-(sp)                        ; visible
    move.w#0,-(sp)                            ; value
    move.w#0,-(sp)                            ; min
    move.w#1,-(sp)                            ; max
    move.w#1,-(sp)                            ; check box proc id
    move.l #0,-(sp)                           ; refcon not used
; Call NewControl
    _NewControl
    lea        AidControl,a3
    move.l (sp)+,(a3)                         ; store the result
; Set up the controls bounding rectangle.
    lea        TRect,a4
    move.w#TCctlInitY+1*TCctlFHeight,(a4)    ; store y coord
    move.w#TCctlInitX,2(a4)                  ; store x coord
    move.w#TCctlInitY+1*TCctlFHeight+20,4(a4) ; store y coord
    move.w#TCRight,6(a4)                      ; store x coord
; Push parameters for NewControl
    clr.l      -(sp)                          ; NewControl returns a handle
    move.l WDHATCPtr,-(sp)                    ; the window ptr
    pea        TRect                          ; the rectangle bounding the control
    pea        'Input Attenuation'           ; title
    move.b #TRUE,-(sp)                        ; visible
    move.w#0,-(sp)                            ; value
    move.w#0,-(sp)                            ; min
    move.w#1,-(sp)                            ; max
    move.w#1,-(sp)                            ; check box proc id
    move.l #0,-(sp)                           ; refcon not used
; Call NewControl
    _NewControl
    lea        IAControl,a3
    move.l (sp)+,(a3)                         ; store the result
; Set up the controls bounding rectangle.
    lea        TRect,a4
    move.w#TCctlInitY+2*TCctlFHeight,(a4)    ; store y coord
    move.w#TCctlInitX,2(a4)                  ; store x coord
    move.w#TCctlInitY+2*TCctlFHeight+20,4(a4) ; store y coord
    move.w#TCRight,6(a4)                      ; store x coord
; Push parameters for NewControl
    clr.l      -(sp)                          ; NewControl returns a handle
    move.l WDHATCPtr,-(sp)                    ; the window ptr
    pea        TRect                          ; the rectangle bounding the control
    pea        'Output Attenuation'          ; title
    move.b #TRUE,-(sp)                        ; visible

```

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```

        move.w #0, -(sp)           ; value
        move.w #0, -(sp)           ; min
        move.w #1, -(sp)           ; max
        move.w #1, -(sp)           ; check box proc id
        move.l #0, -(sp)           ; refcon not used
; Call NewControl
        _NewControl
        lea         OAControl, a3
        move.l (sp)+, (a3)         ; store the result
; Set up the controls bounding rectangle.
        lea         TRect, a4
        move.w #TCctlInitY+3*TCctlFHeight, (a4) ; store y coord
        move.w #TCctlInitX, 2(a4) ; store x coord
        move.w #TCctlInitY+3*TCctlFHeight+20, 4(a4) ; store y coord
        move.w #TCRight, 6(a4) ; store x coord
; Push parameters for NewControl
        clr.l        -(sp)         ; NewControl returns a handle
        move.l WDHATCPtr, -(sp) ; the window ptr
        pea         TRect         ; the rectangle bounding the control
        pea         'Field Mike' ; title
        move.b #TRUE, -(sp)       ; visible
        move.w #1, -(sp)         ; make Field mike on as the default
        move.w #0, -(sp)         ; min
        move.w #1, -(sp)         ; max
        move.w #2, -(sp)         ; radio button proc id
        move.l #0, -(sp)         ; refcon not used
; Call NewControl
        _NewControl
        lea         FieldControl, a3
        move.l (sp)+, (a3)         ; store the result
; Set up the controls bounding rectangle.
        lea         TRect, a4
        move.w #TCctlInitY+4*TCctlFHeight, (a4) ; store y coord
        move.w #TCctlInitX, 2(a4) ; store x coord
        move.w #TCctlInitY+4*TCctlFHeight+20, 4(a4) ; store y coord
        move.w #TCRight, 6(a4) ; store x coord
; Push parameters for NewControl
        clr.l        -(sp)         ; NewControl returns a handle
        move.l WDHATCPtr, -(sp) ; the window ptr
        pea         TRect         ; the rectangle bounding the control
        pea         'Probe Mike' ; title
        move.b #TRUE, -(sp)       ; visible
        move.w #0, -(sp)         ; value
        move.w #0, -(sp)         ; min
        move.w #1, -(sp)         ; max
        move.w #2, -(sp)         ; radio button proc id
        move.l #0, -(sp)         ; refcon not used
; Call NewControl
        _NewControl
        lea         ProbeControl, a3
        move.l (sp)+, (a3)         ; store the result
; Set up the controls bounding rectangle.
        lea         TRect, a4
        move.w #TCctlInitY+5*TCctlFHeight, (a4) ; store y coord

```

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```

        move.w #TCCtlInitX,2(a4)                ; store x coord
        move.w #TCCtlInitY+5*TCCtlFHeight+24,4(a4) ; store y coord
        move.w #TCCtlInitX+40,6(a4)             ; store x coord
; Push parameters for NewControl
        clr.l      -(sp)                        ; NewControl returns a handle
        move.l     WDHATCPtr,-(sp)              ; the window ptr
        pea        TRect                       ; the rectangle bounding the control
        pea        'Start'                     ; title
        move.b     #TRUE,-(sp)                  ; visible
        move.w     #0,-(sp)                     ; value
        move.w     #0,-(sp)                     ; min
        move.w     #0,-(sp)                     ; max
        move.w     #0,-(sp)                     ; simple button proc id
        move.l     #0,-(sp)                     ; refcon not used
; Call NewControl
        _NewControl
        lea        StartControl,a3
        move.l     (sp)+,(a3)                   ; store the result
        movem.l    (sp)+,d0-d7/a0-a6
        rts

```

TCAddBoxes:

```

        movem.l    d0-d7/a0-a6,-(sp)
        lea        TextHandles,a3
        lea        TextRects,a4
        move.w     #ToneBursts,d4

```

TCABLoop:

```

        cmp.w     #TextBoxes,d4
        beq       TCABDone

```

; TENew

; Get Destination Rect in TRect

```

        lea        TRect,a2
        move.l     (a4),(a2)
        move.l     4(a4),4(a2)

```

; Make it a little smaller

```

        pea        TRect
        move.w     #1,-(sp)
        move.w     #1,-(sp)
        _InsetRect

```

; Call TENew

```

        clr.l      -(sp)                        ; make room for handle result
        pea        TRect                       ; dest rect
        pea        TRect                       ; view rect
        _TENew
        move.l     (sp)+,(a3)+
        lea        8(a4),a4
        add.w     #1,d4
        bra        TCABLoop

```

TCABDone:

```

        lea        TextHandles,a4

```

; Default Tone Burst Is 3

```

        pea        '3'                        ; incorporate the text
        add.l      #1,(sp)                   ; move past the length
        move.l     #1,-(sp)                   ; It's 1 character long

```

```

        move.l (a4)+, -(sp)
        _TEInsert
; Default Rise Time is 309
        pea      '309'
        add.l     #1, (sp)
        move.l #3, -(sp)
        move.l (a4)+, -(sp)
        _TEInsert
; Default Signal On is 2455
        pea      '2455'
        add.l     #1, (sp)
        move.l #4, -(sp)
        move.l (a4)+, -(sp)
        _TEInsert
; Default Fall Time is 309
        pea      '309'
        add.l     #1, (sp)
        move.l #3, -(sp)
        move.l (a4)+, -(sp)
        _TEInsert
; Default Signal Off is 3069
        pea      '3069'
        add.l     #1, (sp)
        move.l #4, -(sp)
        move.l (a4)+, -(sp)
        _TEInsert
; Default Frequency is 2000
        pea      '2000'
        add.l     #1, (sp)
        move.l #4, -(sp)
        move.l (a4)+, -(sp)
        _TEInsert
; Default Attenuation is 20
        pea      '20'
        add.l     #1, (sp)
        move.l #2, -(sp)
        move.l (a4)+, -(sp)
        _TEInsert
        movem.l   (sp)+, d0-d7/a0-a6
        rts

```

; Name: WDHATCIdle
 ; Function: This routine blinks the caret of the active text box. It should be
 ; called each time through your main event loop.
 ; Input: None
 ; Output: None

WDHATCIdle:

```

        movem.l   a0-a6/d0-d7, -(sp)
        lea       TextHandles, a4
        move.w WActive, d4
        bmi       TCINoneActive
        asl.w     #2, d4
        move.l (a4, d4.w), -(sp)
        _TEIdle

```

```
TCINoneActive:
    movem.l    (sp)+,a0-a6/d0-d7
    rts
```

```
; Name:WDHATCKey
; Function: Call WDHATCKey when the TC window is active and a keypress
; event is active.
; Input: The char (from the event's message field) as a word.
; Output: None
```

```
WDHATCKey:
    movem.l    a0-a6/d0-d7,-(sp)
    lea        TextHandles,a4
    move.w     WActive,d4          ; which one is active?
    bmi        TCKNoneActive      ; -1 means none
    asl.w      #2,d4              ; *4 for long offset
    move.w     64(sp),-(sp)        ; push the char
    move.l     (a4,d4.w),-(sp)
    _TEKey
```

```
TCKNoneActive:
    movem.l    (sp)+,a0-a6/d0-d7
; remove parameter from stack
    move.l     (sp),2(sp)          ; move return address
    clr.w      (sp)+              ; remove extra space
    rts
```

```
; Name: WDHATCIS
; Function: This routine returns a Boolean telling whether or not
; the given window pointer is the TC window's pointer.
; Input: A window pointer (passed on the stack)
; Output: a word, TRUE or FALSE (defined in WDHA.hdr) returned on the stack.
; **Note: You do not have to push a word for the result of this routine.
```

```
WDHATCIS:
    movem.l    a4/d4,-(sp)        ; save registers
    move.l     8(sp),a4           ; get return address in a4
    move.l     12(sp),d4          ; get WindowPtr in d4
    cmp.l      WDHATCPtr,d4       ; Was it our window?
    beq        IS10              ; It is
    move.w     #FALSE,14(sp)      ; save result
    bra        IS20
IS10:
    move.w     #TRUE,14(sp)
IS20:
    move.l     a4,10(sp)          ; put return address back
    movem.l    (sp)+,a4/d4        ; restore registers
    tst.w      (sp)+             ; get rid of extra two bytes
    rts                          ; return
```

```
; Name: WDHATCControl
; Function: This routine should be called whenever a mousedown event occurs
; within the contents of the TC Window. It handles the highlighting of the
; proper control buttons, and sends the proper records to the WDHA.
; Input: The mouse location (on the stack), from the event's where field.
; Output: None
WDHATCControl:
```

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```

        movem.l    d0-d7/a0-a6,-(sp)
        move.l     WDHATCPtr,-(sp)      ; WDHATCPtr on stack
; PROCEDURE SetPort (gp: Gr2IPort)
        _SetPort
port
        pea       64(sp)                ; push address of point
        _GlobalToLocal                  ; convert it to the window's coords
; Was it in a control button?
ButtonCheck:
; call FindControl
        clr.w      -(sp)                ; returns a long
        move.l     66(sp),-(sp)         ; push point in local coords
        move.l     WDHATCPtr,-(sp)      ; WDHATCPtr on stack
        pea       WhichControl          ; which one?
        _FindControl
        tst.w      (sp)+                ; pop result
        lea        WhichControl,a4
        tst.l      (a4)                 ; Was it in any of them?
        beq        TBCheck              ; if not try the text boxes
; if it was in a control, call TrackControl
        clr.w      -(sp)                ; returns a word
        move.l     WhichControl,-(sp)    ; WhichControl now has the handle
        move.l     70(sp),-(sp)         ; starting point
        move.l     #0,-(sp)             ; no action proc
        _TrackControl
        tst.w      (sp)+                ; did they change the button?
        beq        NoChan               ; if not then leave
; Was it the Start Button?
        move.l     StartControl,d4
        lea        WhichControl,a4
        cmp.l      (a4),d4
        bne        InvControl           ; if not then forget it
        bsr        WDHATCDoTest         ; otherwise do the test
        bra        NoChan               ; and leave
; invert the control value
InvControl:
        clr.w      -(sp)                ; GetCtlValue returns a word
        move.l     WhichControl,-(sp)
        _GetCtlValue
        move.w      (sp)+,d3             ; now value is in d3
        not.w      d3
        and.w      #1,d3
        move.l     WhichControl,-(sp)
        move.w      d3,-(sp)
        _SetCtlValue
; Was it the Field button?
        move.l     FieldControl,d4
        lea        WhichControl,a4
        cmp.l      (a4),d4
        bne        NotField             ; if not then forget it
; Otherwise invert the Probe mike
        clr.w      -(sp)                ; GetCtlValue returns a word
        move.l     ProbeControl,-(sp)

```


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```

        _GetCtlValue
        move.w(sp)+,d3
        not.w d3
        and.w #1,d3
        move.l ProbeControl,-(sp)
        move.wd3,-(sp)
        _SetCtlValue
        bra      NoChan
; Was it the Probe button?
NotField:
        move.l ProbeControl,d4
        lea      WhichControl,a4
        cmp.l   (a4),d4
        bne      NoChan
; Otherwise invert the Field mike
        clr.w   -(sp)
        move.l FieldControl,-(sp)
        _GetCtlValue
        move.w(sp)+,d3
        not.w d3
        and.w #1,d3
        move.l FieldControl,-(sp)
        move.wd3,-(sp)
        _SetCtlValue
        bra      NoChan

TBCheck:
        lea      TextRects,a4
        move.w#ToneBursts,d4

TBCLoop:
        cmp.w #TextBoxes,d4
        beq      NoChan
        clr.w   -(sp)
        move.l 66(sp),-(sp)
        move.l a4,-(sp)
        _PtInRect
        tst.w  (sp)+
        bne      TBFound
        lea      8(a4),a4
        add.w #1,d4
        bra      TBCLoop

TBFound:
; Deactivate old active box
        lea      TextHandles,a3
        lea      WActive,a4
        move.w(a4),d3
        bmi      TBNoneActive
        asl.w #2,d3
        move.l (a3,d3.w),-(sp)
        _TEDeactivate
TBNoneActive
        move.wd4,(a4)
        asl.w #2,d4
        move.l (a3,d4.w),-(sp)
        _TEActivate

```

; now value is in d3
; invert the status
; turn off Probe button
; if not then forget it
; GetCtlValue returns a word
; now value is in d3
; invert the status
; turn off Probe button
; make room for result.
; push the mouse point.
; the text boxes rectangle.
; Is the point inside.
; If so we've found the right one.
; Otherwise move to next rect.
; increment the counter
; Get old active one
; * 4 for long words
; store new active one
; counter * 4 since long words.
; push the TEHandle

```

        move.l 64(sp),-(sp)                ; push the point
        clr.w  -(sp)                      ; don't extend
        move.l (a3,d4.w),-(sp)            ; push the TEHandle
        _TEClick
NoChan:
        _PenNormal
        movem.l (sp)+,d0-d7/a0-a6
        move.l (sp)+,(sp)                  ; get rid of param
        rts

; Name: TCDrawBoxes
; Function: TCDrawBoxes draws the text box portion of the TC window,
; including the headings and the text boxes themselves.
; Input: None
; Output: None
TCDrawBoxes:
        movem.l d0-d7/a0-a6,-(sp)
        pea      ERect                    ; erase the input portion of the window
        _EraseRect
        lea      TextRects,a4
        lea      TextHandles,a3
        move.w #TCctlInitY+16,d3          ; initial y coord
        DispString #10,d3,Tone burst count?
        pea      0(a4)
        _FrameRect
        pea      ERect
        move.l 0(a3),-(sp)
        _TEUpdate
        add.w #20,d3                      ; move down
        DispString #10,d3,Rise time sample count?
        pea      8(a4)
        _FrameRect
        pea      ERect
        move.l 4(a3),-(sp)
        _TEUpdate
        add.w #20,d3                      ; move down
        DispString #10,d3,Signal on sample count?
        pea      16(a4)
        _FrameRect
        pea      ERect
        move.l 8(a3),-(sp)
        _TEUpdate
        add.w #20,d3                      ; move down
        DispString #10,d3,Fall time sample count?
        pea      24(a4)
        _FrameRect
        pea      ERect
        move.l 12(a3),-(sp)
        _TEUpdate
        add.w #20,d3                      ; move down
        DispString #10,d3,Signal off sample count?
        pea      32(a4)
        _FrameRect
        pea      ERect

```

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```

move.l 16(a3),-(sp)
_TEUpdate
add.w #20,d3 ; move down
DispString #10,d3,Frequency?
pea 40(a4)
_FrameRect
pea ERect
move.l 20(a3),-(sp)
_TEUpdate
add.w #20,d3 ; move down
DispString #10,d3,Atten re max out (dB)?
pea 48(a4)
_FrameRect
pea ERect
move.l 24(a3),-(sp)
_TEUpdate
add.w #20,d3 ; move down
DispValue #10,d3,Power = ,PDecimal
pea
_DrawString
lea KeyBuf,a0
move.l PFract,d0
move.w #0,-(SP) ;Select NumToString
_Pack7
pea KeyBuf
_DrawString
movem.l (sp)+,d0-d7/a0-a6
rts

```

; Name: WDHATCDoTest

; Function: WDHATCDoTest fills the paramrec with the proper values

; initiates the WDHA test by sending the paramrec out via the routine

; wdhatest.

; Input: None

; Output: None

WDHATCDoTest

movem.l d0-d7/a0-a6,-(sp) ; save registers

lea paramrec,a4 ; get the gain/input select word

; generate the gain/input select word

move.w 14(a4),d4 ; get the gain input select word in d0

TCIA: ; set input attenuation bit

clr.w -(sp) ; GetCtlValue returns a word

move.l IAControl,-(sp) ; the handle

_GetCtlValue

tst.w (sp)+

beq TCNoIA

TCDIA:

bset.l #INPUT,d4

bra TCOA

TCNoIA:

bclr.l #INPUT,d4

TCOA:

; set output attenuation bit

clr.w -(sp) ; GetCtlValue returns a word

move.l OAControl,-(sp) ; the handle

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```

        _GetCtlValue
        tst.w  (sp)+
        beq          TCNoOA
TCDoOA:
        bset.l  #OUTPUT,d4
        bra          TCField
TCNoOA:
        bclr.l  #OUTPUT,d4
TCField:
        ; set the field mike bit
        clr.w  -(sp)          ; GetCtlValue returns a word
        move.l  FieldControl,-(sp) ; the handle
        _GetCtlValue
        tst.w  (sp)+
        beq          TCNoField
TCDoField:
        bset.l  #FIELD,d4
        bra          TCProbe
TCNoField:
        bclr.l  #FIELD,d4
TCProbe:
        ; set the probe mike bit
        clr.w  -(sp)          ; GetCtlValue returns a word
        move.l  ProbeControl,-(sp) ; the handle
        _GetCtlValue
        tst.w  (sp)+
        beq          TCNoProbe
TCDoProbe:
        bset.l  #PROBE,d4
        bra          TCSendParams
TCNoProbe:
        bclr.l  #PROBE,d4

TCSendParams:
        move.wd4,14(a4)          ; store the modified gain/input select word.
        lea          paramrec,a0
        bsr          TCCvtBoxes
        bsr          wdhatet
        lea          arg1,a4
        move.l  d6,(a4)          ; put MS in arg1
        pea          arg1
        pea          arg2
        fl2X          ; convert MS to extended in arg2
        move.l  d7,(a4)          ; put SMS in arg1
        pea          arg1
        pea          arg3
        fl2X          ; convert SMS to extended in arg3
        move.l  #8388608,(a4)    ; 2^23
        pea          arg1
        pea          arg4
        fl2X          ; convert 2^23 to extended in arg4
        pea          arg4
        pea          arg2
        fdivx        ; divide MS by 2^23 to move decimal point
        pea          arg4
        pea          arg3

```

```

fdivx      ; divide SMS by 2^23 to move decimal point
pea        two
pea        arg3
fdivx      ; SMS/2
pea        arg2
pea        arg2
fmulx      ; MS^2
pea        arg2
pea        arg3
fsubx      ; E in arg3
lea        arg1,a0
move.l     #4342944,(a0)
pea        arg1
pea        arg2
fl2x       ; get 1000000*10/log base e of 10 in arg2
pea        thousand
pea        arg2
fdivx      ; get three decimal places
pea        thousand
pea        arg2
fdivx      ; now six decimal places
pea        arg3
flnx       ; take log base e of E
pea        arg2
pea        arg3
fmulx      ; now Power = (10 * log base e of E)/(log base e of 10) in arg3
pea        arg3
pea        arg2
fx2x       ; copy arg3 (Power) to arg2
pea        arg2
ftintx     ; Truncate result
pea        arg2
pea        arg3
fsubx      ; Now integer part in arg2, fractional part in arg3
pea        thousand
pea        arg3
fmulx      ; get three decimal places
pea        thousand
pea        arg3
fmulx      ; now six decimal places
pea        arg2
pea        arg1
fx2l       ; convert decimal part to long integer
lea        PDecimal,a0
move.l     arg1,(a0)
pea        arg3
pea        arg1
fx2l       ; convert fractional part to long integer
lea        PFract,a1
move.l     arg1,(a1)
bpl        PResult
tst.l      (a0)
beq        PResult
neg.l      (a1)

```

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; Print Result

PResult:

bsr WDHATCDraw

; Now put the WDHA in either hearing aid state or idle state

clr.w -(sp) ; GetCtlValue returns a word

move.l AidControl, -(sp) ; the handle

_GetCtlValue

tst.w (sp)+

beq TCaidOff

move.w #-1, d0 ; go to hearing aid mode

bra TCSetMode

TCaidOff:

move.w #-100, d0 ; go to idle mode

TCSetMode:

jsr scsiwr ; send mode code to WDHA

movem.l (sp)+, d0-d7/a0-a6 ; restore registers

rts

; Name: TCCvtBoxes

; Function: TCCvtBoxes actually does the work of filling the paramrec by
; converting the text of the text boxes to their appropriate values, and by
; calculating the sine and cosine factors from the specified frequency.

; Input: None

; Output: None

TCCvtBoxes:

movem.l d0-d7/a0-a6, -(sp)

lea TextHandles, a4

move.w #ToneBursts, d4

TCCBLoop:

cmp.w #TextBoxes, d4

beq TCCBDone

move.w d4, d5

asl.w #2, d5 ; *4 for longs

move.l (a4, d5.w), a0 ; get the text handle

_HLock ; Lock the handle

move.l (a0), a2 ; Dereference the handle

move.w 60(a2), d6 ; get teLength

lea NumBuf, a6

move.b d6, (a6) ; store the length of the string

clr.l -(sp) ; make room for the result.

move.l a0, -(sp) ; get the text

_TEGetText

move.l (sp)+, a3 ; get it in a3

move.l a3, a0

_HLock ; lock the handle

move.l (a0), a0 ; Dereference the handle, move src in a0

lea NumBufT, a1 ; Destination is NumBufT

move.w d6, d0 ; BlockMove expects length in d0

ext.l d0 ; expects a long

_BlockMove

lea NumBuf, a0

move.w #1, -(SP)

_Pack7 ; StringToNum puts result in d0

lea offsets, a1

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```

move.b (a1,d4.w),d1 ; get offset in paramrec of this entry
ext.w d1 ; make it a word.
lea paramrec,a0 ; get paramrec base address
move.wd0,(a0,d1.w) ; store the value.
move.l a3,a0 ; Unlock the text handle
_HUnlock
move.l (a4,d5.w),a0 ; Unlock the TEHandle
_HUnlock
add.w #1,d4 ; go to next box.
bra TCCBLoop
TCCBDone:
; Now compute the slope delta values which are 16384/sample count
lea paramrec,a4
move.l #16384,d0
move.w2(a4),d1 ; first do the rise time slope delta
beq RTSZero
divu d1,d0
move.wd0,4(a4)
bra FTSDelta
RTSZero:
move.w#$7FFF,4(a4)
FTSDelta:
move.l #16384,d0
move.w8(a4),d1 ; now do the fall time slope delta
beq FTSSZero
divu d1,d0
move.wd0,10(a4)
bra TCCCalcTrig
FTSSZero:
move.w#$7FFF,10(a4)
TCCCalcTrig:
; Now send the parameters to the WDHA
move.wFreq,d0
lea arg1,a1
move.wd0,(a1)
pea arg1
pea arg3 ; arg3 will hold fp frequency
FI2X ;convert from integer to extended fp
; Compute burst amplitude
move.w Atten,d0
bpl AttenOK
clr.w d0
AttenOK:
neg.w d0
lea arg1,a0
move.w d0,(a0) ; store Atten from max output (dB) in arg1
pea arg1 ;dB gain
pea arg4 ;fpdB gain
FI2X ;convert from integer to extended fp
pea fp20dBe ;20 * log base 10 of e = 8.685889638
pea arg4 ;fpdB gain
fdivx ;db/fp20dbe (result in arg4)
pea arg4
fexpx ;base e exponential (db ratio in arg4)

```

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```

    pea    twoex14      ;scale it *2E14 to convert it to fixed point
    pea    arg4
    fmulx
    pea    arg4
    pea    arg1
    fx2i          ;convert extended to integer
    lea      paramrec,a4
    move.warg1,20(a4)  ; store the burst factor
; compute sine and cosine factors
; first get 2*pi*f/fs in arg5
    pea    arg3          ;frequency
    pea    arg5
    fx2x          ;move arg3 to arg5 (frequency)
    pea    twopi        ;2 pi
    pea    arg5
    fmulx          ;multiply 2 pi times f (result in arg5)
    pea    fp12277      ;sampling frequency is 12277 Hz
    pea    arg5
    fdivx          ;divide by fs (result in arg5)
; Now get cos factor
    pea    arg5
    pea    cosreg
    fx2x          ;move arg5 to cosreg
    pea    cosreg
    fcosx          ;take cosine of cosreg
    pea    twoex15      ;2^15
    pea    cosreg
    fmulx          ;multiply by 2^15
    pea    cosreg
    pea    arg1
    fx2i          ;convert extended to integer
    lea      paramrec,a4
    move.warg1,16(a4)  ;store cosine factor
; Now do sine
    pea    arg5
    pea    sinreg
    fx2x          ;move arg5 to sinreg
    pea    sinreg
    fsinx          ;take sine of sinreg
    pea    fp1p95       ;1.95
    pea    sinreg
    fmulx          ;multiply by 1.95
    pea    twoex14      ;2^14
    pea    sinreg
    fmulx          ;multiply by 2^14
    pea    sinreg
    pea    arg2
    fx2i          ;convert extended to integer
    lea      paramrec,a4
    move.warg2,18(a4)  ;push sine factor
    movem.l    (sp)+,d0-d7/a0-a6
    rts

```

-----WDHATC data declarations-----

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```

WDHATCPtr: DC.L 0 ; WDHATC WindowPtr
AidControl: DC.L 0 ; Hearing Aid On Control
IAControl: DC.L 0 ; Input Attenuation Control
OAControl: DC.L 0 ; Output Attenuation
FieldControl: DC.L 0 ; Field Mike Control
ProbeControl: DC.L 0 ; Probe Mike Control
StartControl: DC.L 0 ; Start Button Control

```

```

; Which Text Edit Record is active?

```

```

WActive: dc.w -1 ; -1 means none are active

```

```

TextHandles:

```

```

dcb.l TextBoxes,0

```

```

paramrec: ;WDHA parameter record for test/calibrate
dc.w 1 ;tone burst count
dc.w 0 ;rise time sample count
dc.w 0 ;rise time slope delta
dc.w 16384 ;signal on sample count
dc.w 0 ;fall time sample count
dc.w 0 ;fall time slope delta
dc.w 16384 ;signal off sample count
dc.w 4224 ;gain/input select word
dc.w 0 ;cosine factor
dc.w 0 ;sine factor
dc.w 32000 ;burst amplitude
dc.w 512 ;probe sample count (currently a constant)
dc.w 32 ;probe sample multiplier (currently a constant)

```

```

; The following are not really a part of the paramrec, but currently must
; follow it for the routine TCCvtBoxes to work properly

```

```

Freq: dc.w 0

```

```

Atten: dc.w 0

```

```

; Power

```

```

PDecimal: dc.l 0

```

```

PFract: dc.l 0

```

```

offsets:

```

```

dc.b 0 ;tone burst count is first entry
dc.b 2 ;rise is second
dc.b 6 ;on count is fourth
dc.b 8 ;fall count is next
dc.b 12 ;off count is seventh
dc.b 26 ;frequency is 14th (not really a parameter)
dc.b 28 ;atten is 15th (not really a parameter)

```

```

TextRects:

```

```

dc.w TCCtlInitY+ToneBursts*20
dc.w TCCtlInitX-88
dc.w TCCtlInitY+ToneBursts*20+20
dc.w TCCtlInitX-20

```

```

dc.w TCCtlInitY+RiseCount*20

```

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```

dc.w  TCctInitX-88
dc.w  TCctInitY+RiseCount*20+20
dc.w  TCctInitX-20

dc.w  TCctInitY+OnCount*20
dc.w  TCctInitX-88
dc.w  TCctInitY+OnCount*20+20
dc.w  TCctInitX-20

dc.w  TCctInitY+FallCount*20
dc.w  TCctInitX-88
dc.w  TCctInitY+FallCount*20+20
dc.w  TCctInitX-20

dc.w  TCctInitY+OffCount*20
dc.w  TCctInitX-88
dc.w  TCctInitY+OffCount*20+20
dc.w  TCctInitX-20

dc.w  TCctInitY+Frequency*20
dc.w  TCctInitX-88
dc.w  TCctInitY+Frequency*20+20
dc.w  TCctInitX-20

dc.w  TCctInitY+Attenuate*20
dc.w  TCctInitX-88
dc.w  TCctInitY+Attenuate*20+20
dc.w  TCctInitX-20

```

WDHATCBounds: ; Bounding rect for window

```

DC.W  TCInitY
DC.W  TCInitX
DC.W  TCInitY+200
DC.W  TCRight

```

ERect: ; Bounding rectangle for part to erase

```

DC.W  TCctInitY-8
DC.W  0
DC.W  TCctInitY+7*TCctIFHeight
DC.W  TCctInitX

```

TRect:

```

DC.L  0
DC.L  0 ;For calculating various rectangles.

```

TPoint: DC.L 0 ;For calculating mouse change.

WhichControl: DC.L 0 ; A control handle, for temporary storage.

NumBuf: DC.B 0 ; Buffer for number conversion (length here)
 NumBufT: DC.B 79,0 ; Text here

KeyBuf: DC.B 80,0

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arg1	dc.b.w	8,0	;integer buffer
arg2	dc.b.w	8,0	;extended floating point buffer
arg3	dc.b.w	8,0	;extended floating point buffer
arg4	dc.b.w	8,0	;extended floating point buffer
arg5	dc.b.w	8,0	;extended floating point buffer
cosreg	dc.b.w	8,0	;room for cosine factor
sinreg	dc.b.w	8,0	;room for sine factor
xacc	dc.b.w	8,0	;extended accumulator
txreg	dc.b.w	8,0	;temporary extended register
pi	dc.w	\$4000,\$c90e,\$5604,\$1893,\$74bc	
twopi	dc.w	\$4001,\$c90e,\$5604,\$1893,\$74bc	
zero	dc.w	\$0000,\$0000,\$0000,\$0000,\$0000	
one	dc.w	\$3fff,\$8000,\$0000,\$0000,\$0000	
fp1p95	dc.w	\$3fff,\$f999,\$9999,\$9999,\$999a	
two	dc.w	\$4000,\$8000,\$0000,\$0000,\$0000	
twoex14	dc.w	\$400d,\$8000,\$0000,\$0000,\$0000	
twoex15	dc.w	\$400e,\$8000,\$0000,\$0000,\$0000	
twoex16	dc.w	\$400f,\$8000,\$0000,\$0000,\$0000	
ten	dc.w	\$4002,\$a000,\$0000,\$0000,\$0000	
hundred	dc.w	\$4005,\$c800,\$0000,\$0000,\$0000	
thousand	dc.w	\$4008,\$fa00,\$0000,\$0000,\$0000	
fp12500	dc.w	\$400c,\$c350,\$0000,\$0000,\$0000	
fp12277	dc.w	\$400c,\$bfd4,\$0000,\$0000,\$0000	
fp20dBe	dc.w	\$4002,\$8af9,\$db22,\$d0e5,\$6042	

```
; WDHATC.hdr
; This file must be included if your program uses the
; WDHA Test/Calibrate window.
XREF WDHATCOpen
XREF WDHATCClose
XREF WDHATCShow
XREF WDHATCHide
XREF WDHATCDraw
XREF WDHATCControl
XREF WDHATCIdle
XREF WDHATCKey
XREF WDHATCIS
XREF WDHATCDoTest
```

```

; file WDGHAFC.Asm
;
;   This file contains two routines which read text files containing
;   numeric expressions, and download the numbers to the digital hearing
;   aid. The routine WDHAFCSet is used in the Aid13 program to download
;   filter tap coefficients to the hearing aid. The routine WDHAsSetFileParams
;   is used to download parameters for the SS15 spectral shaping program.
;   The text files accessed by these routines must contain integer numbers
;   separated by any character which is nonnumeric and not '.' (generally spaces,
;   tabs, or carriage returns). The text files accessed by WDHAFCSet can also
;   contain simple numeric expressions of the form A/B, where A and B are
;   integers.
Include MacTraps.D
Include ToolEquX.D
Include SysEquX.D
Include QuickEquX.D
Include FSEqu.D
Include MDS2:WDHADisk.hdr
Include MDS2:WDHASCSI.hdr

XDEF WDHAFCSet
XDEF WDHAsSetFileParams

; Constants for division
NoDiv EQU 0 ; Haven't seen a '/'
ReadOne EQU 1 ; Read first operand
DoDiv EQU 2 ; Read second operand, so don't division.

; Name: WDHAFCSet
; Function: This routine uses the SFGetFile dialog to get the name of the file
;           from the user, then opens the file, converts it's contents from text form
;           to binary integer form, then downloads it to the hearing aid.
; Input: None
; Output: None
WDHAFCSet:
    movem.l d0-d7/a0-a6,-(sp)
; Do SFGetFile
    move.l #$00480048,-(sp) ; where
    pea "Which Filter Coefficient File?" ; prompt
    move.l #0,-(sp) ; fileFilter procedure
    move.w #-1,-(sp) ; display all types of files
    pea FTypes ; typeList
    move.l #0,-(sp) ; dlgHook
    pea Reply ; SFReply
    move.w #2,-(sp) ; trap to SFGetFile
    _Pack3
; Did they choose a file?
    lea good,a3
    tst.w (a3)
    beq DoneFCSet
; Yes, open it.
    lea fName,a1 ; file name pointer
    bsr DiskOpen
    tst.w d1 ; test ioResult
    bne DoneFCSet

```

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```

; Now d2 has ioRefNum
    move.w #1,d1 ; read one sector
    lea     myBuffer,a1
    bsr     DiskRead
    bsr     DiskClose
; Now convert text buffer to words
    move.w #64,d3 ; d3 will be a counter
    move.w #NoDiv,d6 ; d6 tells if we should divide or not
    lea     myBuffer,a1
    lea     numRec,a2
FCLoop:
    lea     numBuffer,a0
; Convert from text buffer to a string
    clr.w   d4 ; count length of string
FCSLoop:
    move.b (a1)+,d5
    cmp.b   #'/',d5
    bne     FCSNotDiv
    move.w #ReadOne,d6
    bra     FCSDone
FCSNotDiv
    cmp.b   #'-',d5
    beq     FCSGo
    cmp.b   #'0',d5
    blo     FCSDone
    cmp.b   #'9',d5
    bhi     FCSDone
FCSGo:
    add.w   #1,d4
    move.b d5,(a0)+
    bra     FCSLoop
FCSDone:
    lea     numString,a0
    move.b d4,(a0)
    move.w #1,-(SP)
    _Pack7 ;StringToNum - cvt numString to word in d0
    cmp.w   #NoDiv,d6 ; Are we dividing?
    beq     FCSDone2
    cmp.w   #ReadOne,d6 ; Have we read one?
    bne     FCSDone1
    add.w   #1,d3 ; This one won't really count
    move.w #DoDiv,d6 ; Next time we'll divide
    bra     FCSDone2
FCSDone1:
    cmp.w   #DoDiv,d6 ; Should be dividing if we reach here
    bne     FCSDone2
    move.w d0,d1 ; get the divisor in d1
    lea     -2(a2),a2 ; back up the pointer to the first operand
    move.w (a2),d0 ; get the first operand
    ext.l   d0 ; extend dest of divs to long
    divs    d1,d0
    move.w #NoDiv,d6 ; finished this divide
    bra     FCSDone2
FCSDone2:

```

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```

        move.wd0,(a2)+      ;store result
        sub.w  #1,d3
        bne    FCLoop
; Send the coefficients to the WDHA
        lea    numRec,a0
        bsr    SetCoefficients
DoneFCSet:
        movem.l (sp)+,d0-d7/a0-a6
        rts

; Name: WDHASetFileParams
; Function: This routine uses the WDHAGetFile dialog to get the file name
;           from the user, then opens the file, converts it's contents from text form
;           to binary integer form, then downloads it to the hearing aid.
; Input: None
; Output: None
WDHASetFileParams:
        movem.l d0-d7/a0-a6,-(sp)
; Do SFGGetFile
        move.l #S00480048,-(sp) ; where
        pea    'Which Set Params File?' ; prompt
        move.l #0,-(sp) ; fileFilter procedure
        move.w #-1,-(sp) ; display all types of files
        pea    FTypes ; typeList
        move.l #0,-(sp) ; dlgHook
        pea    Reply ; SFReply
        move.w #2,-(sp) ; trap to SFGGetFile
        _Pack3
; Did they choose a file?
        lea    good,a3
        tst.w (a3)
        beq    DoneFileSet
; Yes, open it.
        lea    fName,a1 ; file name pointer
        bsr    DiskOpen
        tst.w d1 ; test ioResult
        bne    DoneFileSet
; Now d2 has ioRefNum
        move.w #3,d1 ; read three sectors
        lea    myBuffer,a1
        bsr    DiskRead
        bsr    DiskClose
; Now convert text buffer to words
        move.w #320,d3 ; d3 will be a counter
        lea    myBuffer,a1
        lea    numRec,a2
FileOuterLoop:
        lea    numBuffer,a0
; Convert from text buffer to a string
        clr.w d4 ; count length of string
FileLoop:
        move.b (a1)+,d5
        cmp.b #'-',d5
        beq    FileGo

```

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```

        cmp.b  #'0',d5
        blo    FileDone
        cmp.b  #'9',d5
        bhi    FileDone
FileGo:
        add.w  #1,d4
        move.b d5,(a0)+
        bra    FileLoop
FileDone:
        lea     numString,a0
        move.b d4,(a0)
        move.w #1,-(SP)
        _Pack7      ;StringToNum - cvt numString to word in d0
        move.w d0,(a2)+ ;store result
        sub.w  #1,d3
        bne    FileOuterLoop
; Send the coefficients to the WDHA
        lea     numRec,a0
        bsr     SetFileParams
DoneFileSet:
        movem.l (sp)+,d0-d7/a0-a6
        rts

Reply:
good:   dc.w    0
copy:   dc.w    0
fType:  dc.w    0
vRefNum dc.w    0
version: dc.w    0
fName:  dcb.b   64,0

FTypes:      dc.l    'TEXT'

numString:   dc.b      0      ; length
numBuffer:   dcb.b    63,0    ; text

numRec:      dcb.w    320,0
myBuffer:    dcb.b   1536,0

```


; WDHAFC.hdr
; This file must be included if your program uses the
; Set Filter Coefficients function.
XREF WDHAFCSet
XREF WDHASetFileParams

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```
;WDHASCSI.Asm
;   This file contains routines for sending records back and forth
;   between the Mac and the WDHA via the SCSI bus interface.
```

```
Include MacTraps.D
Include SysEquX.D
Include ToolEquX.D
Include MDS2:WDHA.hdr
```

```
XDEF  SetParam
XDEF  SetCoefficients
XDEF  SetFileParams
XDEF  wdhatst
XDEF  SCSIInterrogate
```

```
XDEF  SCSIWr
XDEF  SCsIRd
XDEF  SCsIBTst
```

```
;scsi bus bit assignments
abs     equ    1           ;assert data bus
dbs     equ    0           ;deassert data bus
ack     equ    0           ;assert acknowledge line
dck     equ    16          ;deassert acknowledge line
atn     equ    0           ;assert attention line
dtn     equ    2           ;deassert attention line
```

```
;;Set WDHA parameters subroutine
```

```
;calling protocol
```

```
;   lea    paramrec,a0    ;set pointer to set parameter record
;   jsr    SetParam
```

```
SetParam:
```

```
    movem.l    a0-a6/d0-d7,-(sp)    ;save registers
    clr.w      -(sp)
    bsr        SCSIInterrogate
    move.w      (sp)+,d0
    beq        @4
    cmp.w      #-100,d0              ;SS15ID
    beq        @4
    move.l      #8-1,d1              ;set loop counter
    move.w      #-2,d0              ;get -2 mode code (set aid parameters)
    jsr        scsiwr              ;send mode code to WDHA
@1    jsr        ScsiBTst          ;test for WDHA
    beq        @1                  ;ready
@2    move.w      (a0)+,d0          ;get parameter
    jsr        scsiwr              ;send parameter to WDHA
@3    jsr        ScsiBTst          ;test for WDHA.
    beq        @3                  ;ready
    dbra       d1,@2              ;check end of loop
    move.w      (a0)+,d0          ;get last parameter
    jsr        scsiwr              ;send last parameter to WDHA
@4    movem.l      (sp)+,a0-a6/d0-d7    ;restore registers
    rts
```

```
;Set WDHA filter coefficients subroutine
```

```
;calling protocol
```

```
;   lea   corec,a0      ;set pointer to array of coefficients
;   jsr   SetCoefficients
```

```
SetCoefficients:
```

```
    movem.l    a0-a6/d0-d7,-(sp)    ;save registers
    move.w #-4,d0                    ;get -4 mode code (set aid coefficients)
    jsr        scsiwr                ;send mode code to WDHA
@1   jsr        ScsiBTst             ;test for WDHA
    beq        @1                    ;ready
    move.l    #63,d1                 ;set loop counter
@2   move.w(a0)+,d0                  ;get parameter
    jsr        scsiwr                ;send parameter to WDHA
@3   jsr        ScsiBTst             ;test for WDHA
    beq        @3                    ;ready
    sub.w     #1,d1                  ;check end of loop
    bne        @2
    move.w(a0)+,d0                    ;get last parameter
    jsr        scsiwr                ;send last parameter to WDHA
    movem.l    (sp)+,a0-a6/d0-d7    ;restore registers
    rts
```

```
;Set file parameters subroutine
```

```
;calling protocol
```

```
;   lea   filerec,a0      ;set pointer to array of 320 coefficients
;   jsr   SetFileParams
```

```
SetFileParams:
```

```
    movem.l    a0-a6/d0-d7,-(sp)    ;save registers
    move.w #-5,d0                    ;get -5 mode code (set aid coefficients)
    jsr        scsiwr                ;send mode code to WDHA
@1   jsr        ScsiBTst             ;test for WDHA
    beq        @1                    ;ready
    move.l    #319,d1                ;set loop counter
@2   move.w(a0)+,d0                  ;get parameter
    jsr        scsiwr                ;send parameter to WDHA
@3   jsr        ScsiBTst             ;test for WDHA
    beq        @3                    ;ready
    sub.w     #1,d1                  ;check end of loop
    bne        @2
    move.w(a0)+,d0                    ;get last parameter
    jsr        scsiwr                ;send last parameter to WDHA
    move.w #-1,d0                    ;get -1 mode code (hearing aid mode)
    jsr        scsiwr                ;send mode code to WDHA
    movem.l    (sp)+,a0-a6/d0-d7    ;restore registers
    rts
```

```
; WDHA test subroutine
```

```
;calling protocol
```

```
;   lea   paramrec,a0      ;set pointer to set parameter record
;   jsr   wdhtest
```

```
; upon exit:
```

```
; d6 has the mean sum
```

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; d7 has the square mean sum
wdhatest:

```

    movem.l    a0-a6/d0-d5,-(sp)    ;save registers
    move.w #-3,d0                    ;get -3 mode code (test/calibrate)
    jsr        scsiwr                ;send mode code to WDHA
@1    jsr        ScsiBTst            ;test for WDHA
    beq        @1                    ;ready
    move.l #13,d1 ;set loop counter (do all but last)
@2    move.w(a0)+,d0                ;get parameter
    jsr        scsiwr                ;send parameter to WDHA
    subq.b #1,d1
    bne        @2                    ;check end of loop

```

; read probe sample

```

@4    jsr        ScsiBTst
    beq        @4                    ;test for WDHA bit

```

; read mean sum

```

    clr.l d0
    jsr        scsiwr                ;write dummy to wdha
    jsr        scsird                ;read high 16 bits
    move.wd0,d6                      ;store in d6
    swap d6                          ;get it in high word
    clr.l d0
    jsr        scsiwr                ;write dummy to wdha
    jsr        scsird                ;read low 9 bits
    move.wd0,d6                      ;store in d6
    asl.w #7,d6                      ;shift it left to the most sig word.
    asr.l #7,d6                      ;shift the whole thing right.

```

; read the mean square sum

```

    clr.l d0
    jsr        scsiwr                ;write dummy to wdha
    jsr        scsird                ;read high 16 bits
    move.wd0,d7                      ;store in d7
    swap d7                          ;get it in most sig word.
    clr.l d0
    jsr        scsiwr                ;write dummy to wdha
    jsr        scsird                ;read low 9 bits
    move.wd0,d7                      ;store in d7
    asl.w #7,d7                      ;shift it left to the most sig word.
    asr.l #7,d7                      ;shift the whole thing right.
    movem.l    (sp)+,a0-a6/d0-d5    ;restore registers

```

; Name: SCSIWr

; Function: Send the 16 bit integer in d0 to the hearing aid via the SCSI bus.

; Input: d0 contains the word to write.

; Output: None

SCSIWr:

```

    movem.l    d0-d3,-(SP)
    move.b #abs+dck+dtn,$580011      ;assert data bus
    move.w #1,d2                      ;set the
    roxr.w #1,d2                      ;extend bit
    move.w #17-1,d2                  ;set loop counter
@1:   roxl.w #1,d0                    ;move in next bit
    move.wd0,d1                      ;copy d0

```

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```

and.w #1,d1 ;mask ls bit
move.b d1,$580001 ;write to output data bus
move.b #abs+ack+dtm,$580011 ;assert acknowledge (clock into wdha)
move.b #abs+dck+dtm,$580011 ;deassert acknowledge (clock into wdha)
dbra d2,@1 ;loop counter
move.w #1000,d3 ;write delay
@2 dbra d3,@2
move.b #dbs+dck+dtm,$580011 ;deassert data bus and all
movem.l (SP)+,d0-d3
rts

```

; Name: SCSIRead

; Function: Read a word from the SCSI bus in register d0.

; Input: None

; Output: d0 contains the word read

```

SCSIRead: movem.l d1-d3,-(SP)
move #16-1,d2 ;set loop counter
move.b #dbs+dck+dtm,$580011 ;deassert data bus and all
@1: asl.w #1,d0 ;shift
move.b $580000,d1 ;read data bus
move.b #dbs+atn+dck,$580011 ;assert attention (clock out wdha)
and.w #2,d1 ;mask input bit (bit 1)
asr.w #1,d1 ;put in position 0
add.w d1,d0 ;add bit to data
move.b #dbs+dtm+dck,$580011 ;deassert attention (clock out wdha)
move.w #250,d3 ;deassert-assert delay
@2 dbra d3,@2 ;loop counter
dbra d2,@1
movem.l (SP)+,d1-d3
rts

```

; Test SCSI read bit (Bit 1). Returns with d0 = 0 or 2

SCSIBtst:

; If the mouse button is pressed then stop communication

```

movem.l a0-a1/d0-d2,-(sp) ; save registers
clr.w -(sp)
_Button
tst.w (sp)+
bne StopCom
movem.l (sp)+,a0-a1/d0-d2
move.b #dbs+dck+dtm,$580011 ;deassert data bus and all
move.b $580000,d0 ;read SCSI bus
and.w #2,d0 ;mask position 1
rts

```

; If the button is pressed during communication we set the hearing aid
; to idle and return to the main loop. Note that extra parameters may
; be left on the stack from the routines which called SCSIBtst.

StopCom:

```

move.w #-5,d0
bsr SCSIWr
bsr SCSIWr
movem.l (sp)+,a0-a1/d0-d2 ; Restore registers
clr.l (sp)+ ; Pop SCSIBtst return address

```

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bra EventLoop

```

; Name: SCSIInterrogate
; Function: Interrogate the hearing aid to determine which program it is running,
;           returning the program identifier code that the hearing aid sends back.
;           If the hearing aid does not respond within a certain timeout period, the
;           routine returns with zero as the result.
; Input: None
; Output: The program code (on the stack)
; ***Note: The user should push a word for the result.

```

SCSIInterrogate:

```

    movem.l    d0-d7/a0-a6,-(sp)
    move.w #-10,d0                ;interrogate WDHA for program type
    bsr    SCSIWr
    clr.w   d0
    move.w #20000,d7
@1    sub.w   #1,d7
    beq     @2
    jsr     ScsiBTst              ;test for WDHA
    beq     @1                   ;ready
@2    jsr     scsird              ;read high 16 bits into d0
    move.w d0,64(sp)
    move.w #-1,d0                ;set hearing aid mode
    bsr     SCSIWr
    movem.l    (sp)+,d0-d7/a0-a6
    rts

```

; WDHASCSI.hdr

XREF SetParam
XREF SetCoefficients
XREF SetFileParams
XREF SCSIInterrogate
XREF wdhatest

XREF SCSIWr
XREF SCSIrD
XREF SCsIBTst

PROBE EQU 9
FIELD EQU 12
INPUT EQU 7
OUTPUT EQU 10

;WDHADisk.asm file

```

Include      FSEqu.D
Include MacTraps.D ; Use System and ToolBox traps
Include      ToolEquX.D ; Use ToolBox equates
Include      SysEquX.D
Include      QuickEquX.D

```

```

XDEF DiskCreate
XDEF DiskRead
XDEF DiskWrite
XDEF DiskEject
XDEF DiskOpen
XDEF DiskClose
XDEF DiskSetFPos
XDEF DiskSetEOF
XDEF DiskSetFInfo

```

```

ioNamePtr    equ    18 ;not included in .d files
ioFVersNum   equ    26 ;not included in .d files
ioMisc       equ    ioRefNum+4 ;not included in .d files

```

DiskRead:

```

;assumes d2 contains ioRefNum
;assumes d1 contains number of 512 byte sectors to read
;assumes a1 points to the buffer to fill
;returns with a0 pointing to parameter block on stack
;and with ioResult in d0
;the number of bytes actually read is returned in d3 (long)

```

```

        moveq #ioVQEISize/2 - 1,d0
@1:     clr.w  -(sp) ;make room on stack for
        dbra  d0,@1 ;for parameter block
        move.l sp,a0 ;set A0 for file manager call

```

```

        move.w d2,ioRefNum(a0) ;and to access parameters in block
        mulu   #512,d1 ;multiply number of sectors by 512
        move.l d1,ioReqCount(a0) ;sectors required
        divu   #512,d1 ;restore d1
        move.l a1,ioBuffer(a0)
        _Read
        move.l ioActCount(a0),d3
        add    #ioVQEISize,SP
        rts

```

DiskWrite:

```

;assumes d2 contains ioRefNum
;assumes d1 contains number of 512 byte sectors to write
;assumes a1 points to the buffer to write
;returns with ioResult in d0
;and a0 pointing to parameter block on stack

```


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```

        moveq #ioVQEISize/2 - 1,d0
@1:     clr.w  -(sp)                ;make room on stack for
        dbra  d0,@1                ;for parameter block
        move.l sp,a0              ;set A0 for file manager call

        move.wd2,ioRefNum(a0)      ;and to access parameters in block
        mulu  #512,d1              ;sectors to write * 512 = bytes
        move.l d1,ioReqCount(a0)   ;blocks of 512 bytes required
        divu  #512,d1              ;restore d1
        move.l a1,ioBuffer(a0)
        _Write
        add   #ioVQEISize,SP
        rts

```

DiskSetFPos:

```

;assumes d2 contains ioRefNum
;assumes d1 contains sector number to position at.
;returns with ioResult in d0
;and a0 pointing to parameter block on stack

```

```

        moveq #ioVQEISize/2 - 1,d0
@1:     clr.w  -(sp)                ;make room on stack for
        dbra  d0,@1                ;for parameter block
        move.l sp,a0              ;set A0 for file manager call

        move.wd2,ioRefNum(a0)      ;and to access parameters in block
        move.w#1,ioPosMode(a0)    ;0 at current position
                                      ;1 relative to beginning of media
                                      ;3 relative to current position

        mulu  #512,d1
        move.l d1,ioPosOffset(a0) ;blocks of 512 bytes required
        divu  #512,d1
        _SetFPos
        add   #ioVQEISize,SP
        rts

```

DiskClose:

```

;assumes d2 contains ioRefNum
;returns with ioResult in d0
;and a0 pointing to parameter block on stack

```

```

        moveq #ioVQEISize/2 - 1,d0
@1:     clr.w  -(sp)                ;make room on stack for
        dbra  d0,@1                ;for parameter block
        move.l sp,a0              ;set A0 for file manager call
                                      ;and to access parameter block

        move.wd2,ioRefNum(a0)      ;ioRefNum in d2 from open routine
        _close
        add   #ioVQEISize,SP
        rts

```

; d3 contains the drive number to eject

DiskEject:

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```

        moveq #ioVQEISize/2 - 1,d0
@1:      clr.w  -(sp)
        dbra   d0,@1
        move.l sp,a0
        move.w #-5,ioRefNum(a0)
        move.w d3,ioDrvNum(a0)
        move.w #ejectCode,csCode(a0)
        _Eject
        add    #ioVQEISize,SP
        rts

```

DiskCreate:

;assumes a1 pointing to file name buffer
;returns with a0 pointing to parameter block on stack
;d3 contains the drive number to create the file on.

```

        moveq #ioVQEISize/2 - 1,d0
@1:      clr.w  -(sp)
        dbra   d0,@1
        move.l sp,a0
                                ;set A0 for file manager call
                                ;and to access parameter block
        move.l a1,ioNamePtr(a0) ;put name pointer in parameter block
        move.b #0,ioFVersNum(a0) ;version number. always use zero
                                ;per page II-81, inside mac
        move.w d3,ioVRefNum(a0) ;drive #
        _Create
        add    #ioVQEISize,SP
        rts

```

DiskOpen:

;assumes a1 pointed to file name buffer
;returns with a0 pointing to parameter block on stack
;ioRefNum in d2 and ioResult in d1
;upon return d3 contains the drive number the file was found on

```

        moveq #ioVQEISize/2 - 1,d0
@1:      clr.w  -(sp)
        dbra   d0,@1
        move.l sp,a0
                                ;set A0 for file manager call
                                ;and to access parameter block
        move.l a1,ioNamePtr(a0) ;put name pointer in parameter block
        move.b #0,ioFVersNum(a0) ;version number. always use zero
                                ;per page II-81, inside mac
        move.w #2,ioVRefNum(a0) ;external drive
        _Open
        move.w #2,d3             ;external drive
        move.w ioRefNum(a0),d2   ;save ioRefNum of file in d2
        move.w ioResult(a0),d1  ;get io result
        beq    DOpenGood
        move.w #1,ioVRefNum(a0) ;internal drive
        _Open
        move.w #1,d3             ;internal drive

```

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```

        move.w ioRefNum(a0),d2          ;save ioRefNum of file in d2
        move.w ioResult(a0),d1         ;get io result
DOpenGood:
        add.l   #ioVQEISize,SP
        rts

DiskSetEOF:
        ;assumes d2 contains ioRefNum
        ;assumes d1 contains position to position at (a long).
        ;returns with ioResult in d0
        ;and a0 pointing to parameter block on stack

        moveq   #ioVQEISize/2 - 1,d0
@1:      clr.w   -(sp)                  ;make room on stack for
        dbra    d0,@1                  ;for parameter block
        move.l   sp,a0                 ;set A0 for file manager call

        move.w d2,ioRefNum(a0)         ;and to access parameters in block
        move.w #1,ioPosMode(a0)       ;0 at current position
                                          ;1 relative to beginning of media
                                          ;3 relative to current position
        move.l   d1,ioMisc(a0)         ;blocks of 512 bytes required
        _SetEOF
        move.w ioResult(a0),d0         ;get io result
        add.l   #ioVQEISize,SP
        rts

DiskSetFinfo:
        ;assumes a1 pointing to file name buffer
        ;assumes d6 contains file creator
        ;assumes d7 contains file type
        ;d3 contains the drive number to create the file on.
        ;returns with a0 pointing to parameter block on stack
        movem.l   d0-d7/a0-a6,-(sp)
        moveq   #ioVQEISize/2 - 1,d0
@1:      clr.w   -(sp)
        dbra    d0,@1
        move.l   sp,a0                 ;set A0 for file manager call
                                          ;and to access parameter block

        move.l   sp,a4
        move.l   a1,ioNamePtr(a0)     ;put name pointer in parameter block
        move.b   #0,ioFVersNum(a0)    ;version number. always use zero
                                          ;per page 11-81, inside mac
        move.w d3,ioVRefNum(a0)       ;drive #
        _GetFileInfo                   ;get file info
        move.l   a4,a0
        move.l   d7,32(a0)
        move.l   d6,36(a0)
        _SetFileInfo
        add.l   #ioVQEISize,SP
        movem.l   (sp)+,d0-d7/a0-a6
        rts

```

```
; WDHADisk.hdr
; This file must be included if your program uses the disk commands.
XREF  DiskCreate
XREF  DiskRead
XREF  DiskWrite
XREF  DiskEject
XREF  DiskOpen
XREF  DiskClose
XREF  DiskSetFPos
XREF  DiskSetEOF
XREF  DiskSetFInfo
```

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Claims:

WHAT IS CLAIMED IS:

1. A hearing aid comprising:
 - a microphone for producing an input signal in response to sound;
 - a plurality of channels connected to a common
 - 5 output, each channel comprising:
 - a filter with preset parameters for receiving the input signal and for producing a filtered signal;
 - a channel amplifier responsive to the filtered
 - 10 signal for producing a channel output signal;
 - means for establishing a channel threshold level for the channel output signal; and
 - means, responsive to the channel output signal and the channel threshold level, for increasing
 - 15 the gain setting of the channel amplifier up to a predetermined limit when the channel output signal falls below the channel threshold level and for decreasing the gain setting of the channel amplifier when the channel output signal
 - 20 rises above the channel threshold level,
 - wherein the channel output signals are combined to produce an adaptively compressed and filtered output signal; and
 - a transducer for producing sound as a function
 - 25 of the adaptively compressed and filtered output signal.

2. The circuit of claim 1 wherein the increasing and decreasing means in each of the channels comprises means for increasing the gain setting of the channel amplifier in increments having a magnitude dp and
5 for decreasing the gain setting of the channel amplifier in decrements having a magnitude dm .

3. The circuit of claim 2 wherein the increasing and decreasing means in each of the channels further comprises:

a comparator for producing a control signal as a
5 function of the level of the channel output signal being greater or less than the channel threshold level;

a gain register for storing a gain setting; and
an adder responsive to the control signal for increasing the gain setting in the gain register by dp
10 when the channel output signal falls below the channel threshold level and for decreasing the gain setting in the gain register by a negative value corresponding to dm when the channel output signal rises above the channel threshold level, wherein the channel amplifier is
15 responsive to the gain register for setting the gain of the channel amplifier as a function of the stored gain setting.

4. The circuit of claim 1 wherein the channel amplifier comprises a two stage amplifier, the first stage having a variable gain and the second stage having a predetermined gain.

5. The circuit of claim 1 wherein the filters in the channels have preset filter parameters for selectively altering the input signal over substantially all of the audible frequency range.

6. The circuit of claim 1 wherein each filter in the channels has preset filter parameters for selectively passing the input signal over a predetermined range of audible frequencies, each filter substantially
5 attenuating any of the input signal not occurring in the predetermined range.

7. The circuit of claim 1 further comprising:
a second channel amplifier responsive to the filtered signal for producing a second channel output signal; and

5 means for programming the gain setting of the second channel amplifier as a function of the gain setting of the first channel amplifier,

wherein the second channel output signals are combined for producing a programmably compressed and
10 filtered output signal and wherein the transducer produces sound as a function of the programmably compressed and filtered output signal.

8. The circuit of claim 7 wherein the programming means comprises means for varying the gain setting of the second channel amplifier as a function of a power of the gain setting of the first channel amplifier.

9. The circuit of claim 7 wherein the filters in the channels have preset filter parameters for selectively altering the input signal over substantially all of the audible frequency range.

10. The circuit of claim 7 wherein each filter in the channels has preset filter parameters for selectively passing the input signal over a predetermined range of audible frequencies, each filter substantially
5 attenuating any of the input signal not occurring in the predetermined range.

11. A hearing aid comprising:
a microphone for producing an input signal in
response to sound;
an amplifier for receiving the input signal and
5 for producing an output signal;
means for establishing a threshold level for the
output signal;
means, responsive to the output signal and the
threshold level, for increasing the gain setting of the
10 amplifier up to a predetermined limit in increments having
a magnitude δp when the output signal falls below the
threshold level and for decreasing the gain setting of the
amplifier in decrements having a magnitude δm when the
output signal rises above the threshold level, and
15 a transducer for producing sound as a function
of the output signal.

12. The circuit of claim 11 wherein the
increasing and decreasing means further comprises:
a comparator for producing a control signal as a
function of the level of the output signal being greater
5 or less than the threshold level;
a gain register for storing a gain setting; and
an adder responsive to the control signal for
increasing the gain setting in the gain register by δp
when the output signal falls below the threshold level and
10 for decreasing the gain setting in the gain register by a
negative value corresponding to δm when the output signal
rises above the threshold level, wherein the amplifier is
responsive to the gain register for setting the gain of
the amplifier as a function of the stored gain setting.

13. The circuit of claim 11 wherein the
amplifier comprises a two stage amplifier, the first stage
having a variable gain and the second stage having a
predetermined gain.

14. The circuit of claim 11 further comprising:
a second amplifier responsive to the input
signal for producing a second output signal; and
means for programming the gain setting of the
5 second amplifier as a function of the gain setting of the
first amplifier,
wherein the transducer produces sound as a
function of the second output signal.

15. The circuit of claim 14 wherein the
programming means comprises means for varying the gain of
the second amplifier as a function of a power of the gain
setting of the first amplifier.

16. An adaptive compressing and filtering
circuit comprising a plurality of channels connected to a
common output, each channel comprising:
a filter with preset parameters for receiving an
5 input signal in the audible frequency range for producing
a filtered signal;
a channel amplifier responsive to the filtered
signal for producing a channel output signal;
means for establishing a channel threshold level
10 for the channel output signal; and
means, responsive to the channel output signal
and the channel threshold level, for increasing the gain
of the channel amplifier up to a predetermined limit when
the channel output signal falls below the channel
15 threshold level and for decreasing the gain of the channel
amplifier when the channel output signal rises above the
channel threshold level,
whereby the channel output signals are combined
to produce an adaptively compressed and filtered output
20 signal.

17. The circuit of claim 16 wherein the increasing and decreasing means in each of the channels comprises means for increasing the gain of the channel amplifier in increments having a magnitude dp and for
5 decreasing the gain of the channel amplifier in decrements having a magnitude dm .

18. The circuit of claim 17 wherein the increasing and decreasing means in each of the channels further comprises:

a comparator for producing a control signal as a
5 function of the level of the channel output signal being greater or less than the channel threshold level;
a gain register for storing a gain setting; and
an adder responsive to the control signal for increasing the gain setting in the gain register by dp
10 when the channel output signal falls below the channel threshold level and for decreasing the gain setting in the gain register by a negative value corresponding to dm when the channel output signal rises above the channel threshold level, wherein the channel amplifier is
15 responsive to the gain register for setting the gain of the channel amplifier as a function of the stored gain setting.

19. The circuit of claim 18 wherein the increasing and decreasing means in each channel further comprises a secondary register for storing the values of dp and dm , said adder being responsive to said register
5 for receiving the stored values of dp and dm for increasing and decreasing the stored gain setting in the gain register.

20. The circuit of claim 16 wherein the channel amplifier comprises a two stage amplifier, the first stage having a variable gain and the second stage having a predetermined gain.

21. The circuit of claim 16 further comprising means for producing a timing sequence, wherein the increasing and decreasing means in each channel is enabled in response to the timing sequence for increasing or
5 decreasing the channel amplifier gain during a predetermined portion of the timing sequence.

22. The circuit of claim 16 wherein each channel further comprises means for clipping the channel output signal at a predetermined level for producing an adaptively clipped and compressed channel output signal.

23. The circuit of claim 16 wherein the filters in the channels have preset filter parameters for selectively altering the input signal over substantially all of the audible frequency range.

24. The circuit of claim 16 wherein each filter in the channels has preset filter parameters for selectively passing the input signal over a predetermined range of audible frequencies, each filter substantially
5 attenuating any of the input signal not occurring in the predetermined range.

25. The circuit of claim 16 wherein the filters in each of the channels comprise finite impulse response filters.

26. The circuit of claim 16 wherein each channel further comprises:

a second channel amplifier responsive to the filtered signal for producing a second channel output
5 signal; and

means for programming the gain setting of the second channel amplifier as a function of the gain of the first channel amplifier,

whereby the second channel output signals are
10 combined for producing a programmably compressed and filtered output signal.

27. The circuit of claim 26 wherein the programming means comprises means for varying the gain of the second channel amplifier as a function of a power of the gain setting of the first channel amplifier.

28. The circuit of claim 27 wherein the programming means further comprises a register for storing a power value and wherein the programming means varies the gain of the second channel amplifier as a function of the
5 value derived by raising the gain setting of the first channel amplifier to the power of the stored power value.

29. The circuit of claim 26 wherein the first and second channel amplifiers each comprise a two stage amplifier, the first stage having a variable gain and the second stage having a preset gain.

30. An adaptive gain amplifier circuit comprising:

an amplifier for receiving an input signal in the audible frequency range and producing an output
5 signal;

means for establishing a threshold level for the output signal; and

means, responsive to the output signal and the threshold level, for increasing the gain of the amplifier up to a predetermined limit in increments having a magnitude dp when the output signal falls below the threshold level and for decreasing the gain of the amplifier in decrements having a magnitude dm when the output signal rises above the threshold level, whereby the output signal is compressed as a function of the ratio of dm over dp to produce an adaptively compressed output signal.

31. The circuit of claim 30 wherein the increasing and decreasing means comprises:

a comparator for producing a control signal as a function of the level of the output signal being greater or less than the threshold level;

a gain register for storing a gain setting; and an adder responsive to the control signal for increasing the gain setting in the gain register by dp when the output signal falls below the threshold level and for decreasing the gain setting in the gain register by a negative value corresponding to dm when the output signal rises above the threshold level, wherein the amplifier is responsive to the gain register for setting the gain of the amplifier as a function of the stored gain setting.

32. The circuit of claim 31 wherein the increasing and decreasing means further comprises means for producing a timing sequence, said increasing and decreasing means being enabled in response to the timing sequence for increasing or decreasing the stored gain setting in the gain register during a predetermined portion of the timing sequence.

33. The circuit of claim 31 wherein the increasing and decreasing means further comprises a secondary register for storing the values of dp and dm and wherein the adder is responsive to said register for
5 receiving the stored values of dp and dm for increasing and decreasing the stored gain setting in the gain register.

34. The circuit of claim 31 wherein the amplifier comprises a two stage amplifier, the first stage having a variable gain and the second stage having a predetermined gain.

35. The circuit of claim 30 wherein the amplifier comprises a two stage amplifier, the first stage having a variable gain and the second stage having a predetermined gain.

36. The circuit of claim 30 further comprising means for clipping the adaptively compressed output signal at a predetermined level and for producing an adaptively clipped compressed output signal.

37. A programmable compressive gain amplifier circuit comprising:

a first amplifier for receiving an input signal in the audible frequency range and for producing an
5 amplified signal;

means for establishing a threshold level for the amplified signal;

means, responsive to the amplified signal and the threshold level, for increasing the gain setting of
10 the first amplifier up to a predetermined limit when the amplified signal falls below the threshold level and for decreasing the gain setting of the first amplifier when

the amplified signal rises above the threshold level,
thereby compressing the amplified signal;

- 15 a second amplifier for receiving the input
signal and for producing an output signal; and
 means for programming the gain setting of the
second amplifier as a function of the gain setting of the
first amplifier,
20 whereby the output signal is programmably
compressed.

38. The circuit of claim 37 wherein the
increasing and decreasing means comprises means for
increasing the gain of the first amplifier in increments
having a magnitude dp and for decreasing the gain of the
5 first amplifier in decrements having a magnitude dm .

39. The circuit of claim 38 wherein the
increasing and decreasing means further comprises:
 a comparator for producing a control signal as a
function of the level of the amplified signal being
5 greater or less than the threshold level;
 a gain register for storing a gain setting; and
 an adder responsive to the control signal for
increasing the gain setting in the gain register by dp
when the amplified signal falls below the threshold level
10 and for decreasing the gain setting in the gain register
by a negative value corresponding to dm when the amplified
signal rises above the threshold level, wherein the first
amplifier is responsive to the gain register for setting
the gain of the first amplifier as a function of the
15 stored gain setting.

40. The circuit of claim 39 wherein the
increasing and decreasing means further comprises means
for producing a timing sequence, said increasing and
decreasing means being enabled in response to the timing

- 5 sequence for increasing or decreasing the stored gain setting in the gain register during a predetermined portion of the timing sequence.

41. The circuit of claim 39 wherein the increasing and decreasing means further comprises a secondary register for storing the values of dp and dm and wherein the adder is responsive to said register for
5 receiving the stored values of dp and dm for increasing or decreasing the stored gain setting in the gain register.

42. The circuit of claim 37 wherein the programming means comprises means for varying the gain of the second amplifier as a function of a power of the gain setting of the first amplifier.

43. The circuit of claim 42 wherein the programming means further comprises a register for storing a power value and wherein the programming means varies the gain of the second amplifier as a function of the value
5 derived by raising the gain setting of the first amplifier to the power of the stored power value.

44. The circuit of claim 37 wherein the first and second amplifiers each comprise a two stage amplifier, the first stage having a variable gain and the second stage having a preset gain.

45. The circuit of claim 37 further comprising means for clipping the adaptively compressed output signal at a predetermined level and for producing an adaptively clipped compressed output signal.

46. A hearing aid comprising:

a microphone for producing an input signal in response to sound;

5 a filter with variable parameters for receiving the input signal and for producing an adaptively filtered signal;

an amplifier for receiving the adaptively filtered signal and for producing an adaptively filtered output signal;

10 means for detecting a characteristic of the input signal;

means, responsive to the detecting means, for varying the parameters of the variable filter and for varying the gain of the amplifier as functions of the
15 detected characteristic; and

a transducer for producing sound as a function of the adaptively filtered output signal.

47. The circuit of claim 46 wherein the varying means comprises a memory for storing a table of characteristic values, related filter parameters, and related amplifier gain values; wherein the memory is
5 responsive to the detecting means for recalling a filter parameter and amplifier gain value as functions of the detected characteristic; and wherein the variable filter is responsive to the memory for setting the parameters of the variable filter as a function of the recalled filter
10 parameter and wherein the amplifier is responsive to the memory for setting the gain of the amplifier as a function of the recalled gain value.

48. A hearing aid comprising:

a microphone for producing an input signal in response to sound;

5 a plurality of channels connected to a common output, each comprising a filter with preset parameters

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for receiving the input signal and for producing a filtered signal and an amplifier responsive to the filtered signal for producing a channel output signal;

a second filter with preset parameters
10 responsive to the input signal for producing a characteristic signal;

a detector responsive to the characteristic signal for producing a control signal, the detector including means for programming the time constant of the
15 detector;

means responsive to the detector for producing a log value representative of the control signal; and

a memory for storing a preselected table of log values and gain values; wherein the memory is responsive
20 to the log value producing means for selecting a gain value for each of the amplifiers in the channels as a function of the produced log value, and wherein each of the amplifiers in the channels is responsive to the memory for separately varying the gain of the respective
25 amplifier as a function of the respective selected gain value; and

a transducer for producing sound as a function of the combined channel output signals.

49. An adaptive compressing and filtering circuit comprising;

a plurality of channels connected to a common output, each channel comprising a filter with preset
5 parameters for receiving an input signal in the audible frequency range and for producing a filtered signal, and a channel amplifier responsive to the filtered signal for producing an output signal;

means for detecting a characteristic of the
10 output signal in each channel; and

means, responsive to the detecting means, for separately varying the gains of the amplifiers in each of

the channels to compress the output signal as a function of the detected characteristic,

15 whereby the output signals in the channels are combined to produce an adaptively compressed and filtered output signal.

50. The circuit of claim 49 wherein the channel amplifiers each comprise a two stage amplifier, wherein the first stage has a predetermined gain for defining an operating range for the respective channel and the second
5 stage has a variable gain responsive to the varying means.

51. The circuit of claim 49 wherein the varying means further comprises means for sequentially modifying the gain setting in each of the channels from first to last as a function of the level of the input signal.

52. The circuit of claim 49 wherein the filters in the channels have preset filter parameters for selectively altering the input signal over substantially all of the audible frequency range.

53. The circuit of claim 49 wherein each filter in the channels has preset filter parameters for selectively passing the input signal over a predetermined range of audible frequencies, each filter substantially
5 attenuating any of the input signal not occurring in the predetermined range.

54. The circuit of claim 49 wherein the filters in each of the channels comprise finite impulse response filters.

55. The circuit of claim 49 wherein the detecting means further comprises means for establishing a channel threshold level for the output signal in each channel; and wherein the varying means further comprises
5 means for increasing the gain of each amplifier up to a predetermined limit when the output signal in the respective channel falls below the channel threshold level and for decreasing the gain of said amplifier when said output signal rises above said channel threshold level.

56. The circuit of claim 55 wherein the increasing and decreasing means comprises means for increasing the gains of the amplifiers in increments having a magnitude dp and for decreasing the gains of the
5 amplifiers in decrements having a magnitude dm .

57. The circuit of claim 56 wherein the increasing and decreasing means further comprises:

a comparator for producing a control signal as a function of the level of the output signal in a channel
5 being greater or less than the channel threshold level;

a gain register for storing a gain setting for each amplifier; and

an adder responsive to the control signal for increasing the gain setting for an amplifier by dp when
10 the output signal in the respective channel falls below the channel threshold level and for decreasing the gain setting for the amplifier by a negative value corresponding to dm when the output signal in the channel rises above the channel threshold level, wherein said
15 amplifier is responsive to the gain register for setting the gain of said amplifier as a function of said stored gain setting.

58. The circuit of claim 57 wherein the increasing and decreasing means further comprises a secondary register for storing the values of dp and dm , and wherein the adder is responsive to the secondary
5 register for receiving the stored values of dp and dm for increasing and decreasing the stored gain settings in the gain register.

59. The circuit of claim 49 further comprising means for producing a timing sequence, wherein the varying means is enabled in response to the timing sequence for varying the gains of the amplifiers during a predetermined
5 portion of the timing sequence.

60. The circuit of claim 49 wherein each channel further comprises means for clipping the output signal in each channel at a predetermined level for producing an adaptively clipped and compressed output
5 signal.

61. An adaptive filtering circuit comprising:
a plurality of channels connected to a common output, each comprising a filter with preset parameters for receiving an input signal in the audible frequency
5 range for producing a filtered signal and an amplifier responsive to the filtered signal for producing a channel output signal;

a second filter with preset parameters responsive to the input signal for producing a
10 characteristic signal; and

a detector responsive to the characteristic signal for producing a control signal, the detector including means for programming the time constant of the detector;

15 means responsive to the detector for producing a log value representative of the control signal; and

a memory for storing a preselected table of log values and gain values; wherein the memory is responsive to the log value producing means for selecting a gain value for each of the amplifiers in the channels as a function of the produced log value, and wherein each of the amplifiers in the channels is responsive to the memory for separately varying the gain of the respective amplifier as a function of the respective selected gain value, whereby the channel output signals are combined to produce an adaptively filtered output signal.

62. The circuit of claim 61 wherein the filters in the channels have preset filter parameters for selectively altering the input signal over substantially all of the audible frequency range.

63. The circuit of claim 61 wherein each filter in the channels has preset filter parameters for selectively passing the input signal over a predetermined range of audible frequencies, each filter substantially attenuating any of the input signal not occurring in the predetermined range.

64. The circuit of claim 61 wherein the filters in each of the channels comprise finite impulse response filters, and wherein the filter in the detecting means comprises a finite impulse response filter.

65. The circuit of claim 61 wherein the second filter is constituted by one of the filters in one of the channels.

66. An adaptive filtering circuit comprising:
a filter with variable parameters for receiving an input signal in the audible frequency range and for producing an adaptively filtered signal;

5 an amplifier for receiving the adaptively filtered signal and for producing an adaptively filtered output signal;

means for detecting a characteristic of the input signal; and

10 means, responsive to the detecting means, for varying the parameters of the variable filter and for varying the gain of the amplifier as functions of the detected characteristic.

67. The circuit of claim 66 wherein the detecting means comprises;

a filter responsive to the input signal for producing a characteristic signal; and

5 a detector responsive to the characteristic signal for producing a control signal, the detector including means for programming the time constant of the detector; and wherein the varying means comprises means for varying the parameters of the variable filter and for
10 varying the gain of the amplifier as a function of the control signal.

68. The circuit of claim 67 wherein the varying means comprises:

means responsive to the detector for producing a log value representative of the control signal; and

5 a memory for storing a preselected table of log values and related filter parameters and gain values,

said memory being responsive to the log value producing means for selecting a filter parameter and a gain value as a function of the produced log value, said
10 variable filter being responsive to the memory for setting the parameters of the variable filter as a function of the selected filter parameter, and said amplifier being responsive to the memory for setting the gain of the amplifier as a function of the selected gain value.

69. The circuit of claim 66 wherein the varying means comprises a memory for storing a table of characteristic values, related filter parameters, and related amplifier gain values; wherein the memory is
5 responsive to the detecting means for recalling a filter parameter and amplifier gain value as functions of the detected characteristic; and wherein the variable filter is responsive to the memory for setting the parameters of the variable filter as a function of the recalled filter
10 parameter and wherein the amplifier is responsive to the memory for setting the gain of the amplifier as a function of the recalled gain value.

70. The circuit of claim 66 further comprising:
a limiter for receiving the adaptively filtered output signal and producing a limited output signal; and
a second filter with variable parameters for
5 receiving the limited output signal and producing a filtered output signal, wherein the parameters of the second variable filter vary as a function of the parameters of the first variable filter.

71. The circuit of claim 70 wherein the varying means comprises a memory for storing a table of characteristic values, related filter parameters, and related amplifier gain values; wherein the memory is
5 responsive to the detecting means for recalling a filter parameter and amplifier gain value as functions of the detected characteristic; and wherein the first and second variable filters are responsive to the memory for setting the parameters of said filters as a function of the
10 recalled filter parameter and wherein the amplifier is responsive to the memory for setting the gain of the amplifier as a function of the recalled gain value.

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FIG. 1

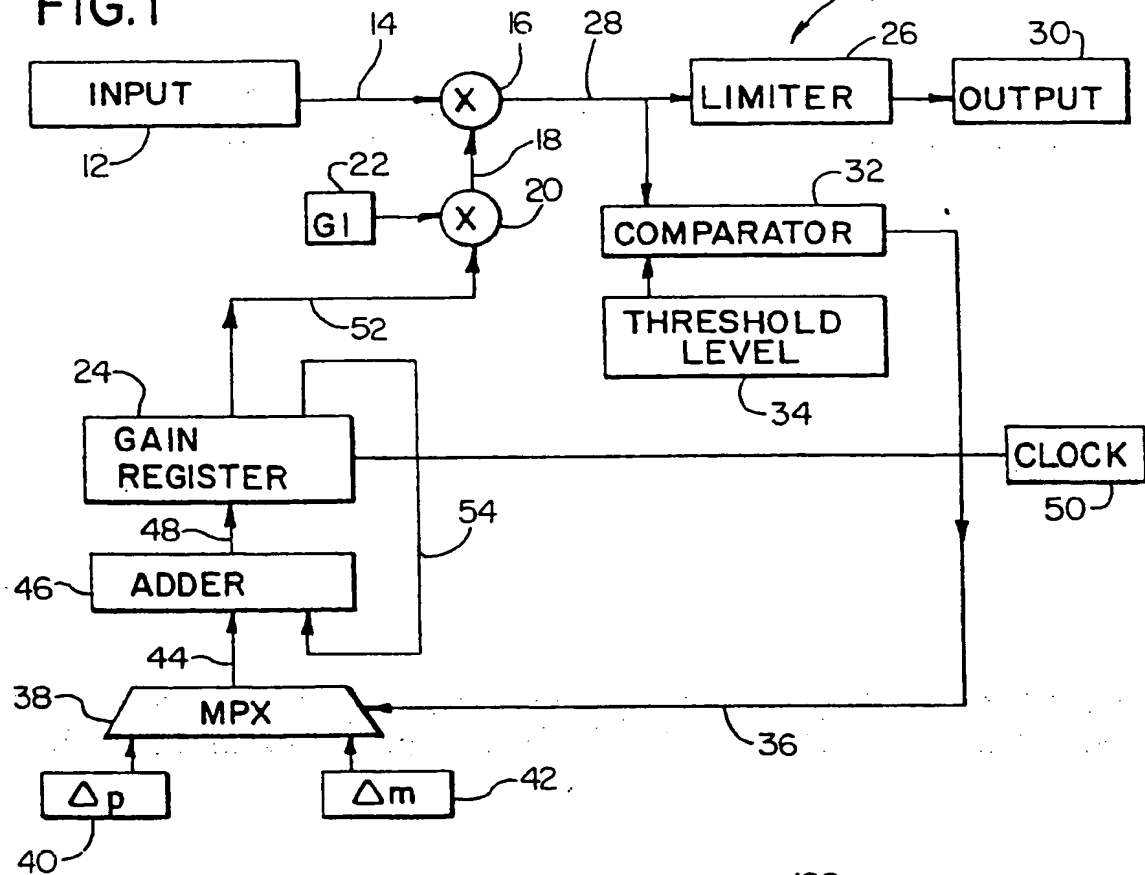
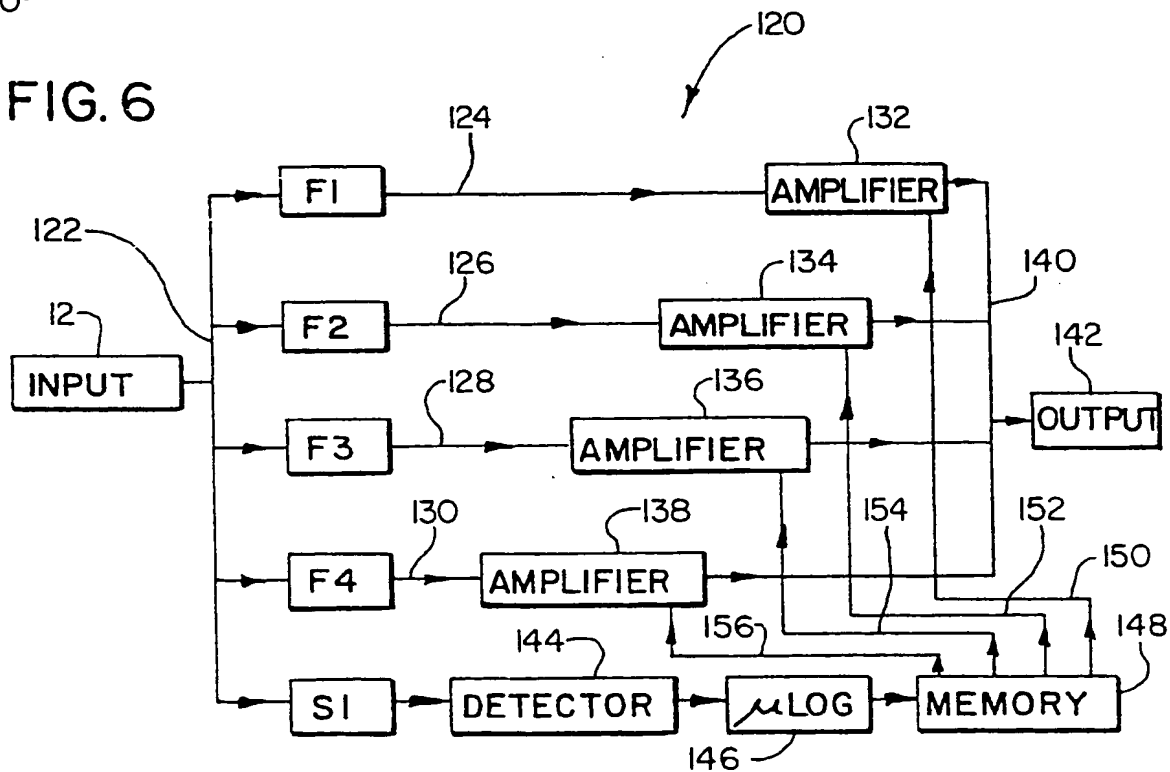
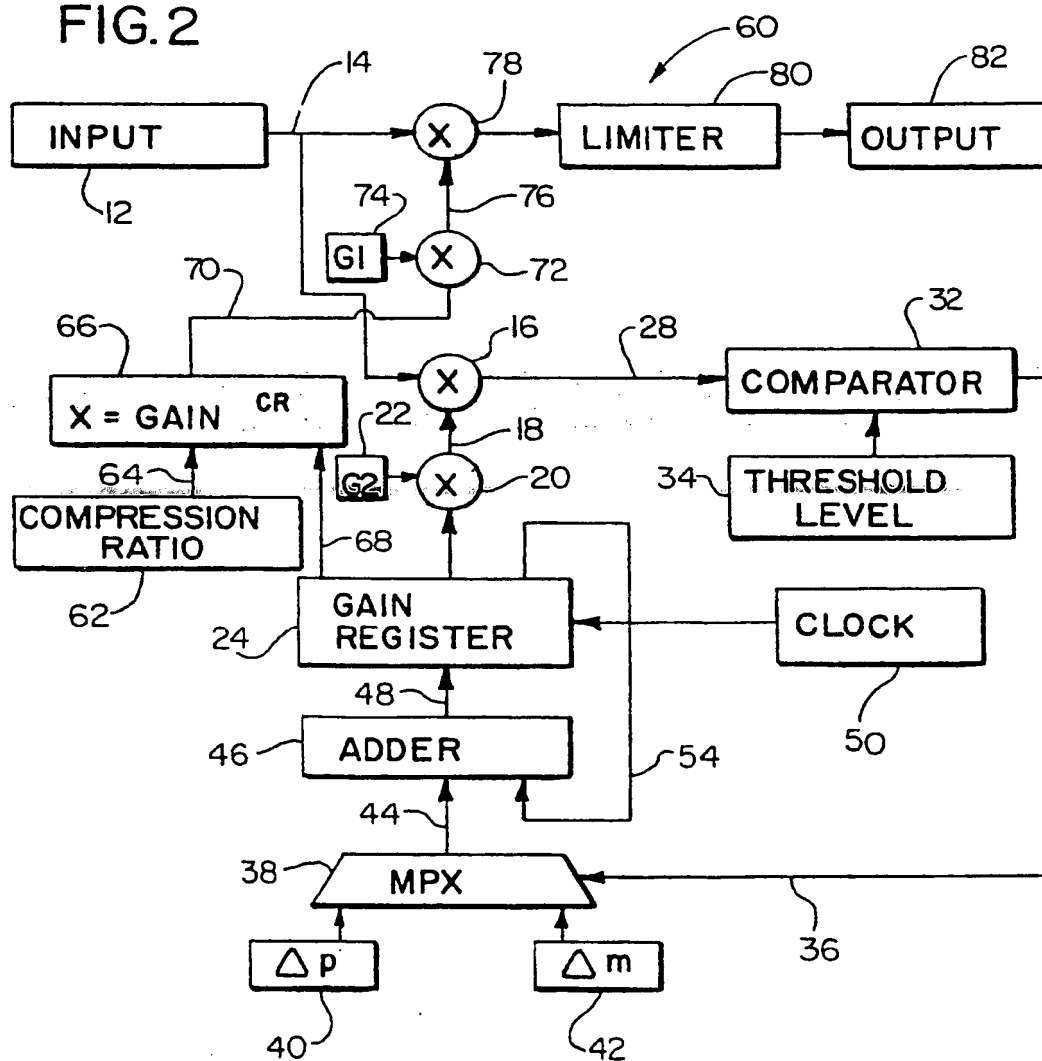


FIG. 6



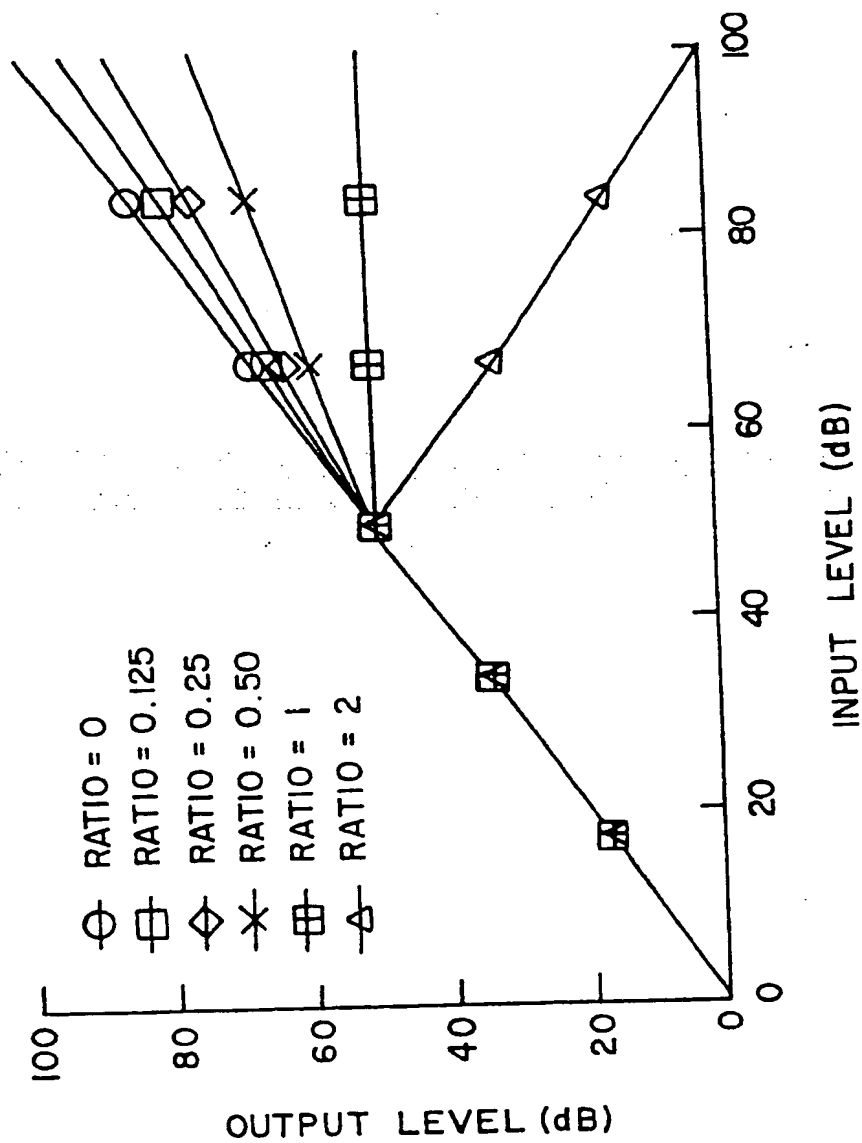
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FIG. 2



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FIG. 3
INPUT/OUTPUT CURVES AS A
FUNCTION OF COMPRESSION RATIO



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FIG. 7

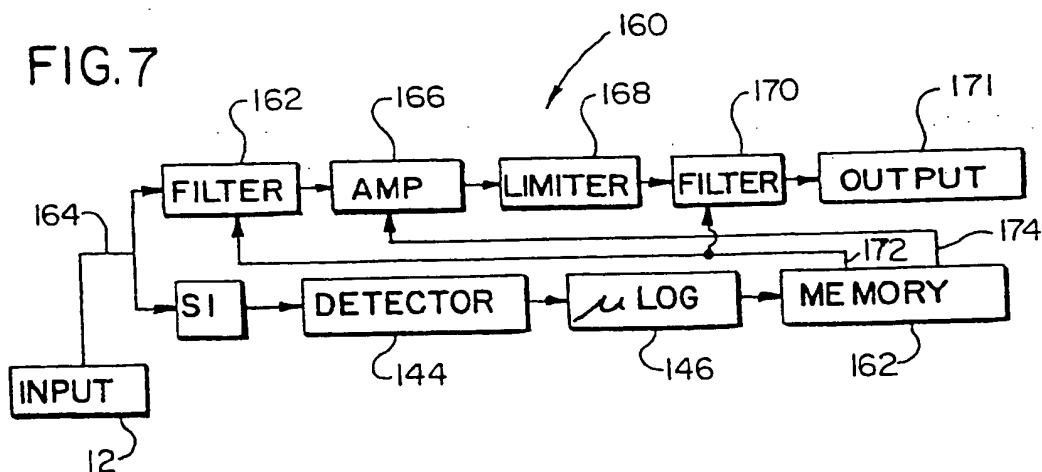


FIG. 8

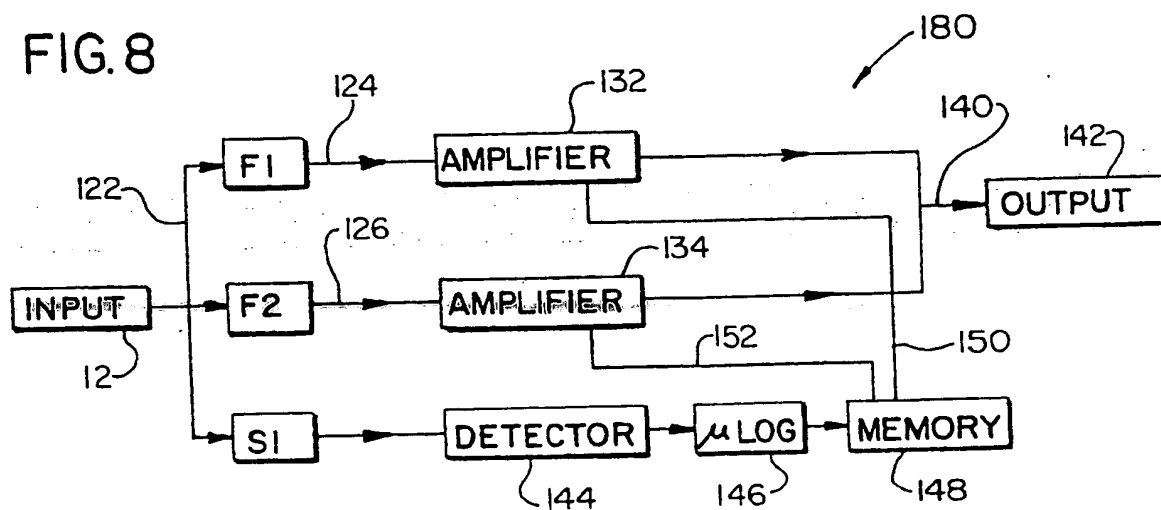
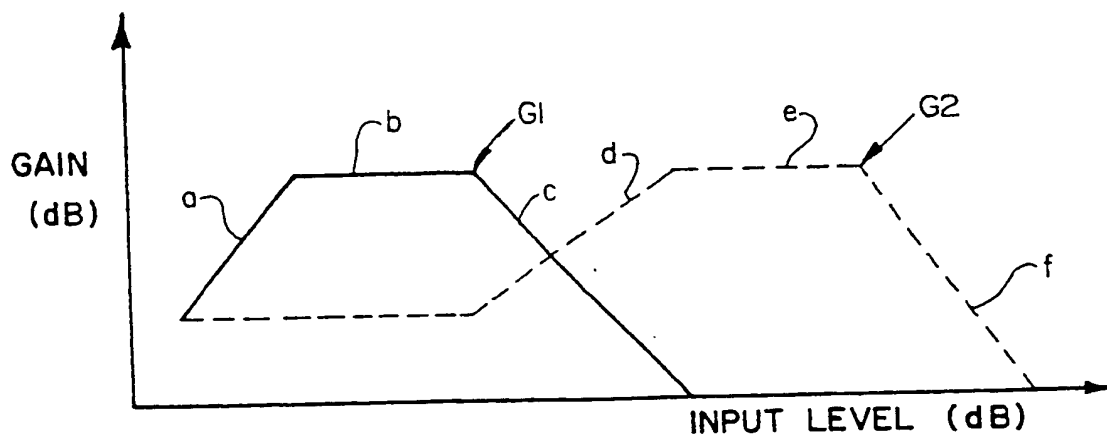


FIG. 9



INTERNATIONAL SEARCH REPORT

Inter onal application No.
PCT/US94/04004

A. CLASSIFICATION OF SUBJECT MATTER

IPC(5) : H04R 25/00

US CL : 381/68.4

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 381/68.4, 68, 68.2, 106; 333/14

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

N/A

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

APS

search terms: filter#, amplifier#, threshold, gain(3a)register, comparator

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US, A, 4,508,940 (STEEGER) 02 April 1985, Fig. 1.	1-2, 5-6, 14-17, 20-25, 30, 36-38, 42, 45, 49, 51-56, 60, 66 and 70
X	US, A, 4,630,302 (KRYTER) 16 December 1986, Figs. 1-2.	11 and 30
X	US, A, 4,680,798 (NEUMANN) 14 July 1987, Fig. 1.	46, 49, 51-56, 59, 66-67 and 69

☐ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

* Special categories of cited documents:	* T	later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
* A* document defining the general state of the art which is not considered to be part of particular relevance	* X*	document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
* E* earlier document published on or after the international filing date	* Y*	document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
* L* document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)	* A*	document member of the same patent family
* O* document referring to an oral disclosure, use, exhibition or other means		
* P* document published prior to the international filing date but later than the priority date claimed		

Date of the actual completion of the international search

02 August 1994

Date of mailing of the international search report

01 SEP 1994

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5706352

INTERNATIONAL SEARCH REPORT

Intern. application No.
PCT/US94/04004**Box I Observations where certain claims were found unsearchable (Continuation of item 1 of first sheet)**

This international report has not been established in respect of certain claims under Article 17(2)(a) for the following reasons:

1. ☐ Claims Nos.:
because they relate to subject matter not required to be searched by this Authority, namely:

2. ☐ Claims Nos.:
because they relate to parts of the international application that do not comply with the prescribed requirements to such an extent that no meaningful international search can be carried out, specifically:

3. ☐ Claims Nos.:
because they are dependent claims and are not drafted in accordance with the second and third sentences of Rule 6.4(a).

Box II Observations where unity of invention is lacking (Continuation of item 2 of first sheet)

This International Searching Authority found multiple inventions in this international application, as follows:

Group I. Claims 1-29 and 46-60 drawn to a hearing aid with adaptive compressing and filtering circuit classified in class 381 subclass 68.4.

Group II. Claims 30-45 drawn to a gain amplifier circuit classified in class 381, subclass 121.

Group III. Claims 61-71 drawn to an adaptive filtering circuit classified in class 364, subclass 724.01.

1. ☒ As all required additional search fees were timely paid by the applicant, this international search report covers all searchable claims.
2. ☐ As all searchable claims could be searched without effort justifying an additional fee, this Authority did not invite payment of any additional fee.
3. ☐ As only some of the required additional search fees were timely paid by the applicant, this international search report covers only those claims for which fees were paid, specifically claims Nos.:

4. ☐ No required additional search fees were timely paid by the applicant. Consequently, this international search report is restricted to the invention first mentioned in the claims; it is covered by claims Nos.:

Remark on Protest

- ☐ The additional search fees were accompanied by the applicant's protest.
☒ No protest accompanied the payment of additional search fees.